Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Inter-Tel 5000 with the Mediant 2000 Gateway using

T1-CAS (In-band DTMF)

By : AudioCodes

Updated Since : 2007-05-28

READ THIS BEFORE YOU PROCEED

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Content

This document describes the configuration required to setup Inter-Tel 5000 and AudioCodes Mediant 2000 using T1 CAS with inband DTMF as the telephony signaling protocol. It also contains the results of the interoperability testing of Microsoft Exchange 2007 Unified Messaging based on this setup.

Intended Audience

This document is intended for Systems Integrators with significant telephony knowledge.

Technical Support

The information contained within this document has been provided by Microsoft partners or equipment manufacturers and is provided AS IS. This document contains information about how to modify the configuration of your PBX or VoIP gateway. Improper configuration may result in the loss of service of the PBX or gateway. Microsoft is unable to provide support or assistance with the configuration or troubleshooting of components described within. Microsoft recommends readers to engage the service of a Microsoft Exchange 2007 Unified Messaging Specialist or the manufacturers of the equipment(s) described within to assist with the planning and deployment of Exchange Unified Messaging.

Microsoft Exchange 2007 Unified Messaging (UM) Specialists

These are Systems Integrators who have attended technical training on Exchange 2007 Unified Messaging conducted by Microsoft Exchange Engineering Team. For contact information, visit here.

Version Information

Date of Modification	Details of Modification
4/6/2007	Initial pass for Inter-Tel configuration

1. Components Information

1.1. PBX or IP-PBX

PBX Vendor	Inter-Tel
Model	5000 series platforms
Software Version	Inter-Tel 5000 v2.1 and greater
Telephony Signaling	T1 CAS with in-band DTMF
Additional Notes	

1.2. VoIP Gateway

Gateway Vendor	Audiocodes
Model	Mediant 2000
Software Version	4.80A.027.002
VoIP Protocol	SIP

1.3. Microsoft Exchange Server 2007 Unified Messaging

Version	RTM

2. Prerequisites

2.1. PBX Requisites

- The Analog Voice Mail Hunt Group premium feature must be enabled in the Inter-Tel system.
- The Inter-Tel 5000 doesn't support the sending of the calling user number for direct calls (i.e., the user calls UM to retrieve voice message). Therefore, when the user dials directly to the Exchange 2007 UM Server, the user hears the general welcome prompt: "Welcome, you are connected to Microsoft Exchange, to access your mailbox, enter your extension.", at which the user is required to enter the user's extension number in addition to the pin number.

2.2. Gateway Requisites

The gateway also supports TLS (in addition to TCP). This provides security by enabling the encryption of SIP packets over the IP network. The gateway supports self-signed certificates as well as Microsoft Windows Certificates Authority (CA) capabilities.

2.3. Cabling Requirements

This integration uses standard RJ-48c cables to connect digital trunks (T1/E1) between the PBX and Mediant 2000 trunk interfaces.

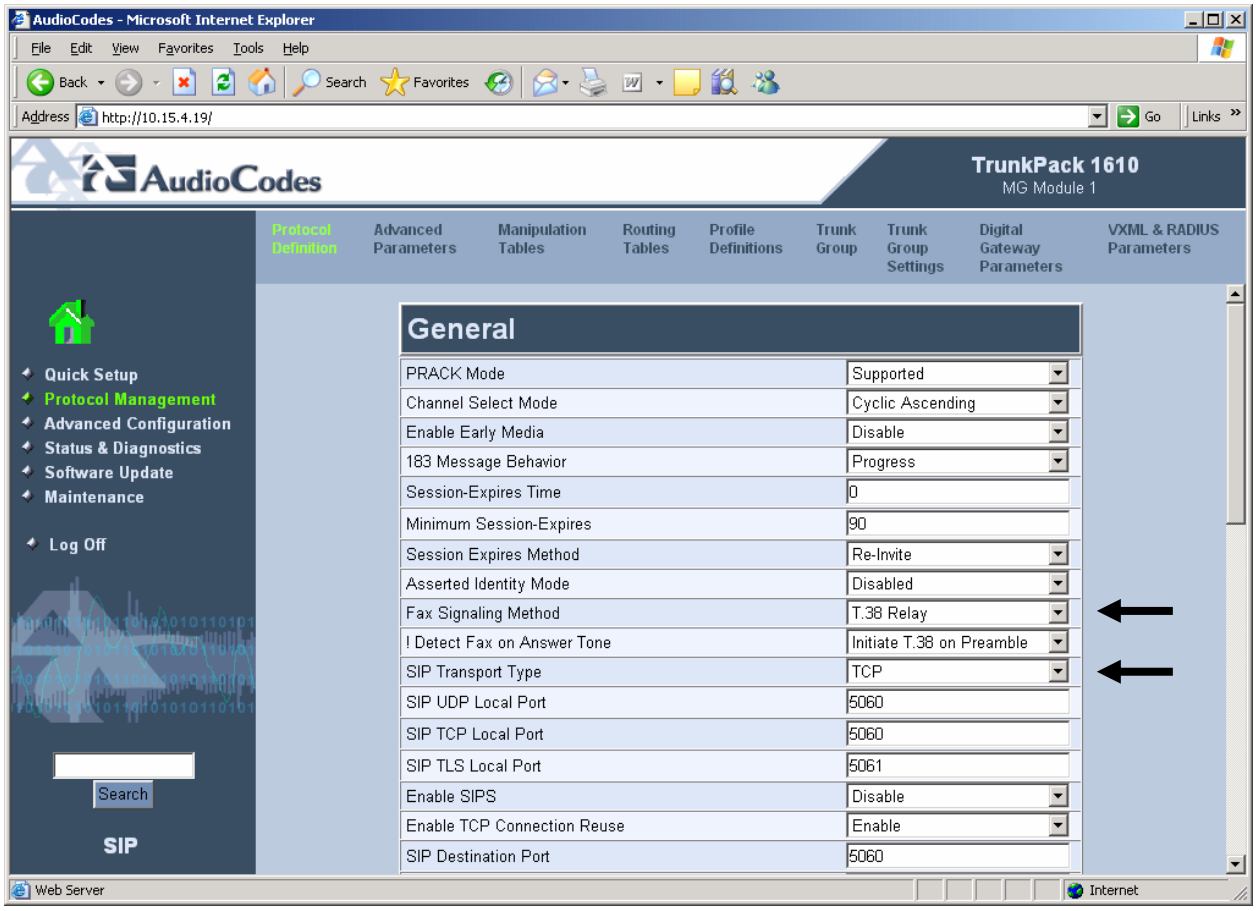
3. Summary and Limitations

☐ A check in this box indicates the UM feature set is fully functional when using the PBX/gateway in question.

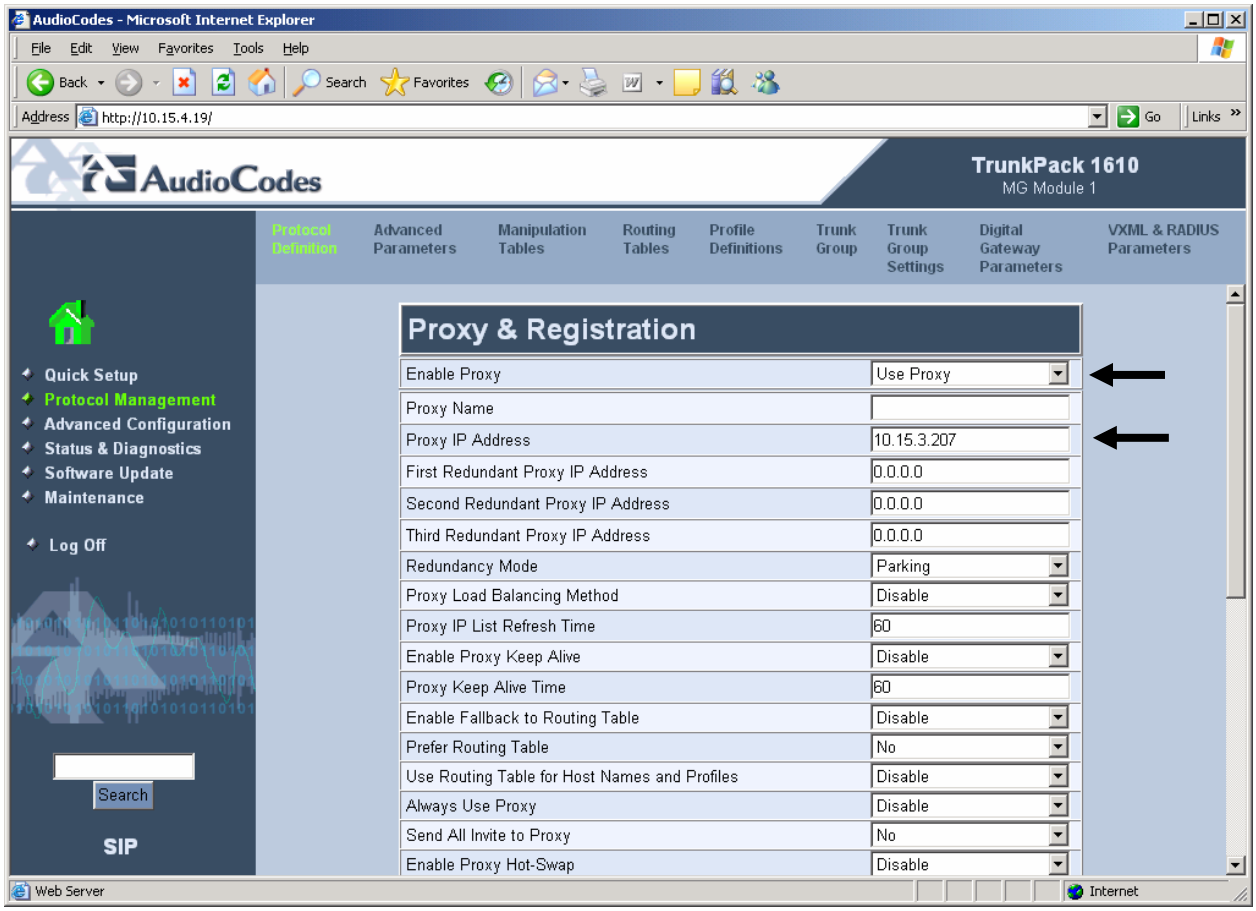
- Since this solution implements T1-CAS, the caller ID of the external caller to UM can't be sent by the PBX. Therefore, the calling party ID is not part of the delivered message.
- When using subscriber access, it's necessary to first enter the mailbox, and then the password.

4. Gateway Setup Notes

Step 1: SIP Environment Setup

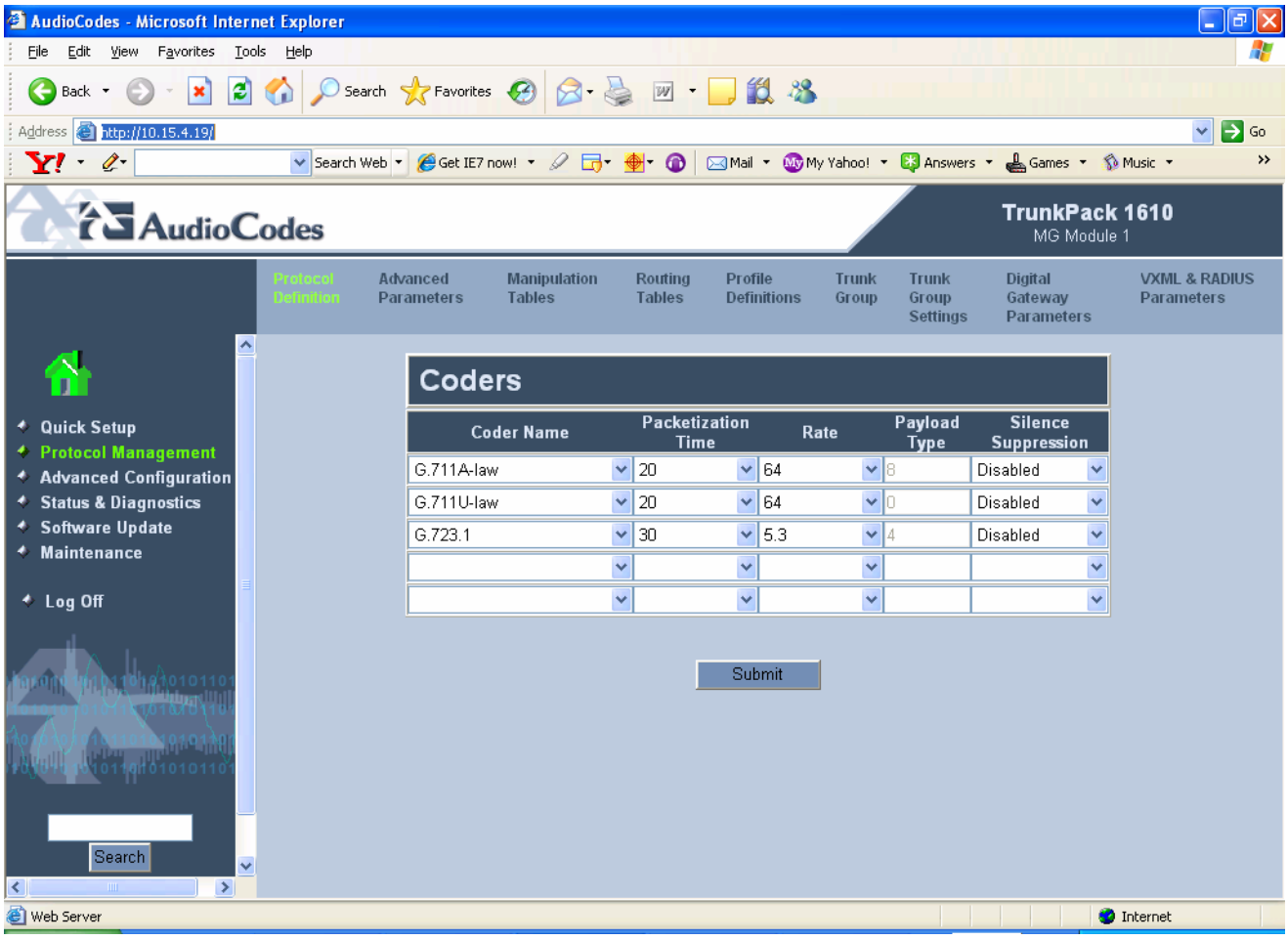


Step 2: Routing Setup

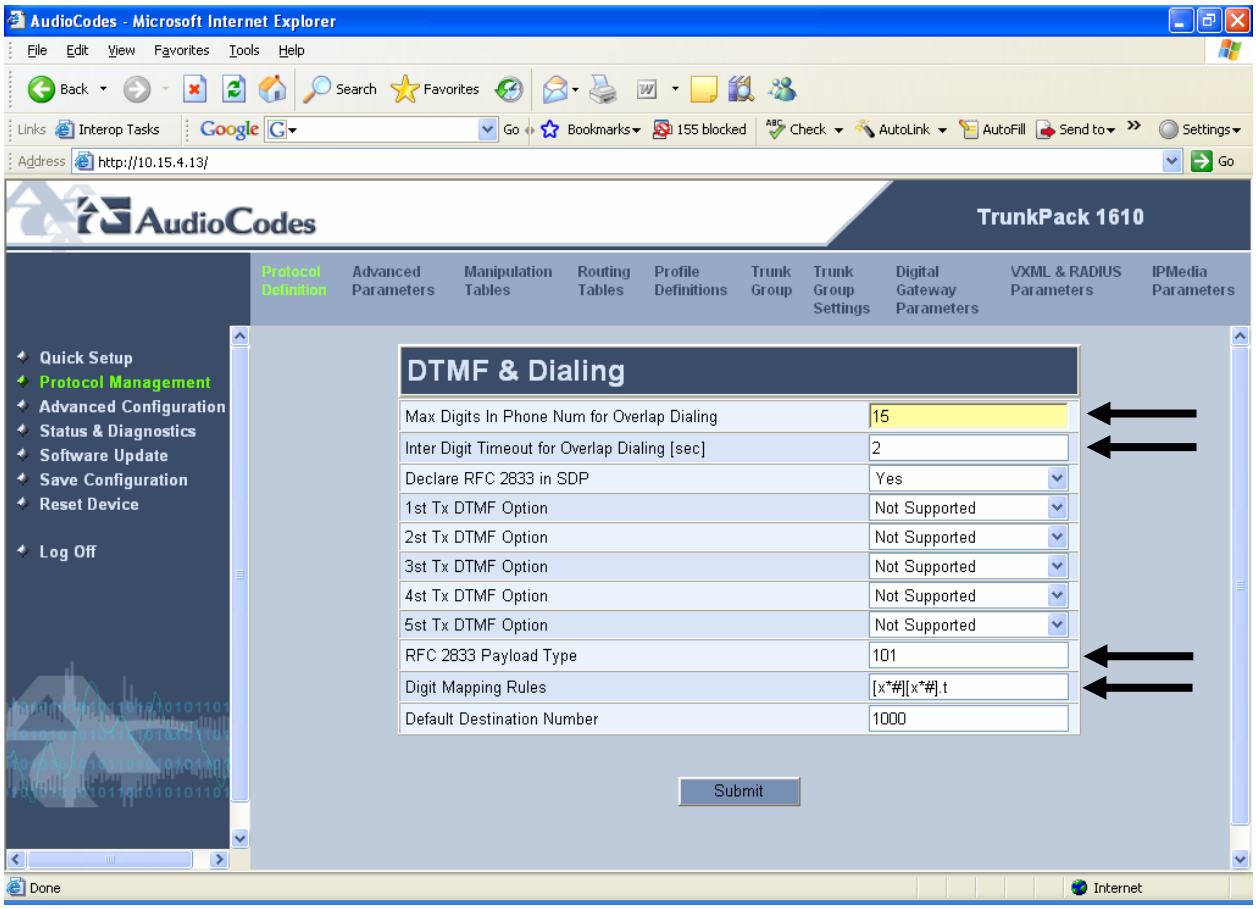


Note: The Proxy IP Address must be one that corresponds to the network environment in which the Microsoft Unified messaging server is installed (For example, 10.15.3.207 or the FQDN of the Microsoft Unified messaging host).

Step 3: Coder Setup



Step 4: Digit Collection Setup

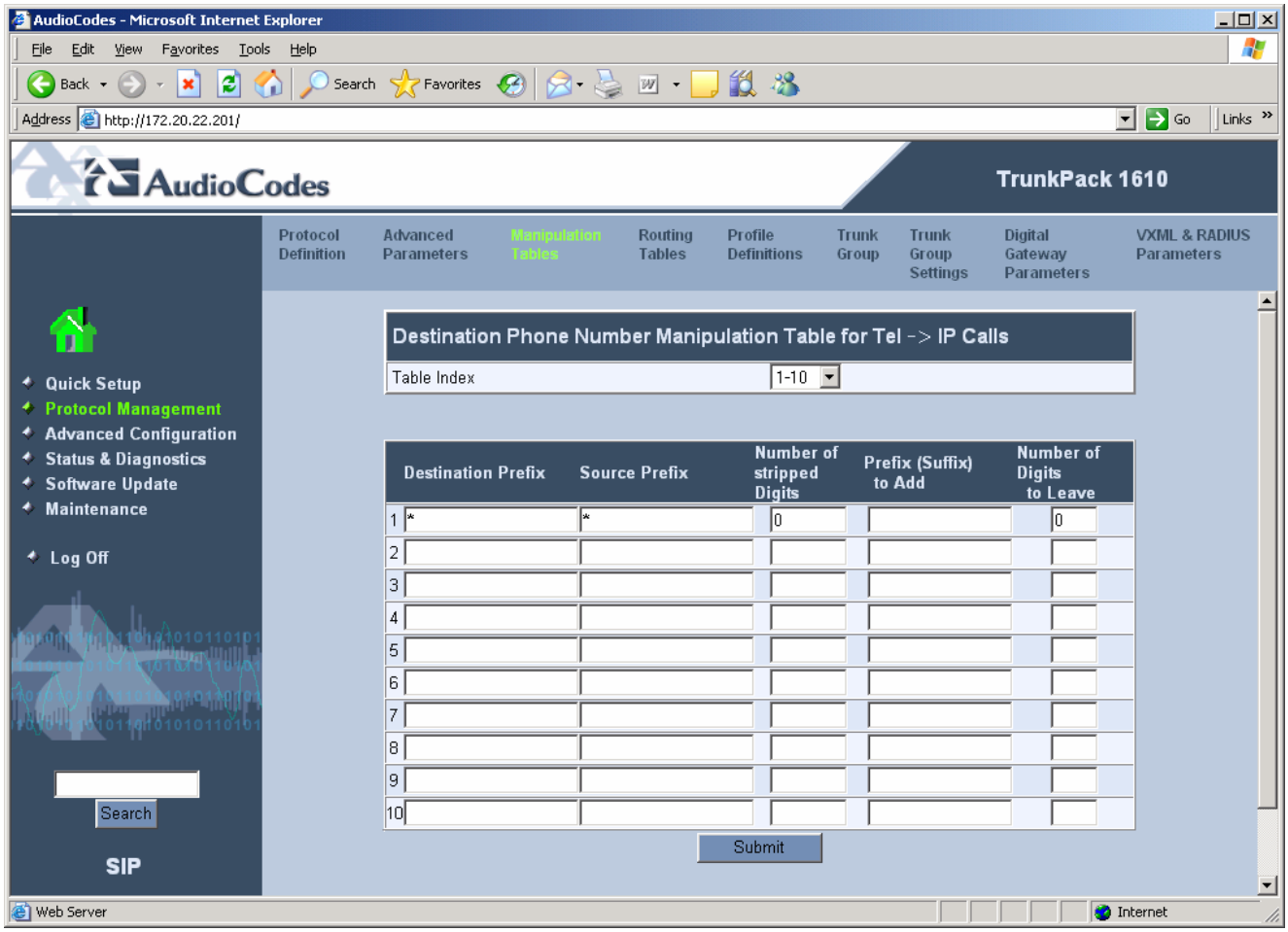


Step 5: Manipulation Routing Setup

There are two options to setup the Manipulation Routing:

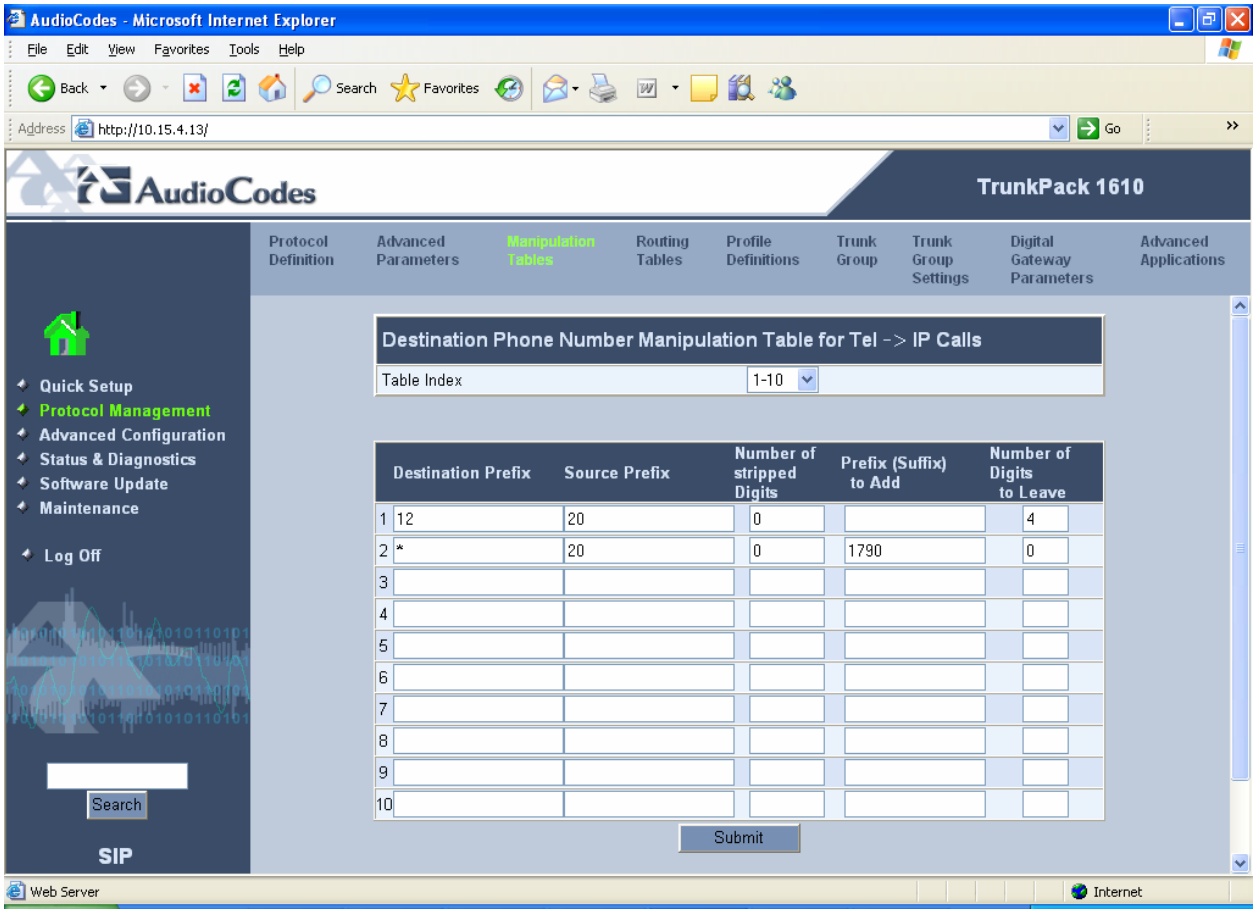
- For general deployments, use the manipulation routing setup as described in Step 5(a).
- For Auto Attendant and Unified Messaging combined solution, use Step 5(a) or Step 5(b), which is an example used for our specific testing environment.

Step 5 (a): Standard Manipulation Routing Setup



Step 5 (b): Manipulation Routing Example

In this example, the dial plan of the Inter-Tel system uses *12xx*. The Source Prefix is the phone number in the Trunk Group Settings step. In this example setup, *1790* is the Auto Attendant pilot number of the Microsoft Unified Messaging Server. The configuration in this table resends inbound calls destined for the Auto Attendant, to Microsoft Unified Messaging with the configured pilot number for the Auto Attendant application.

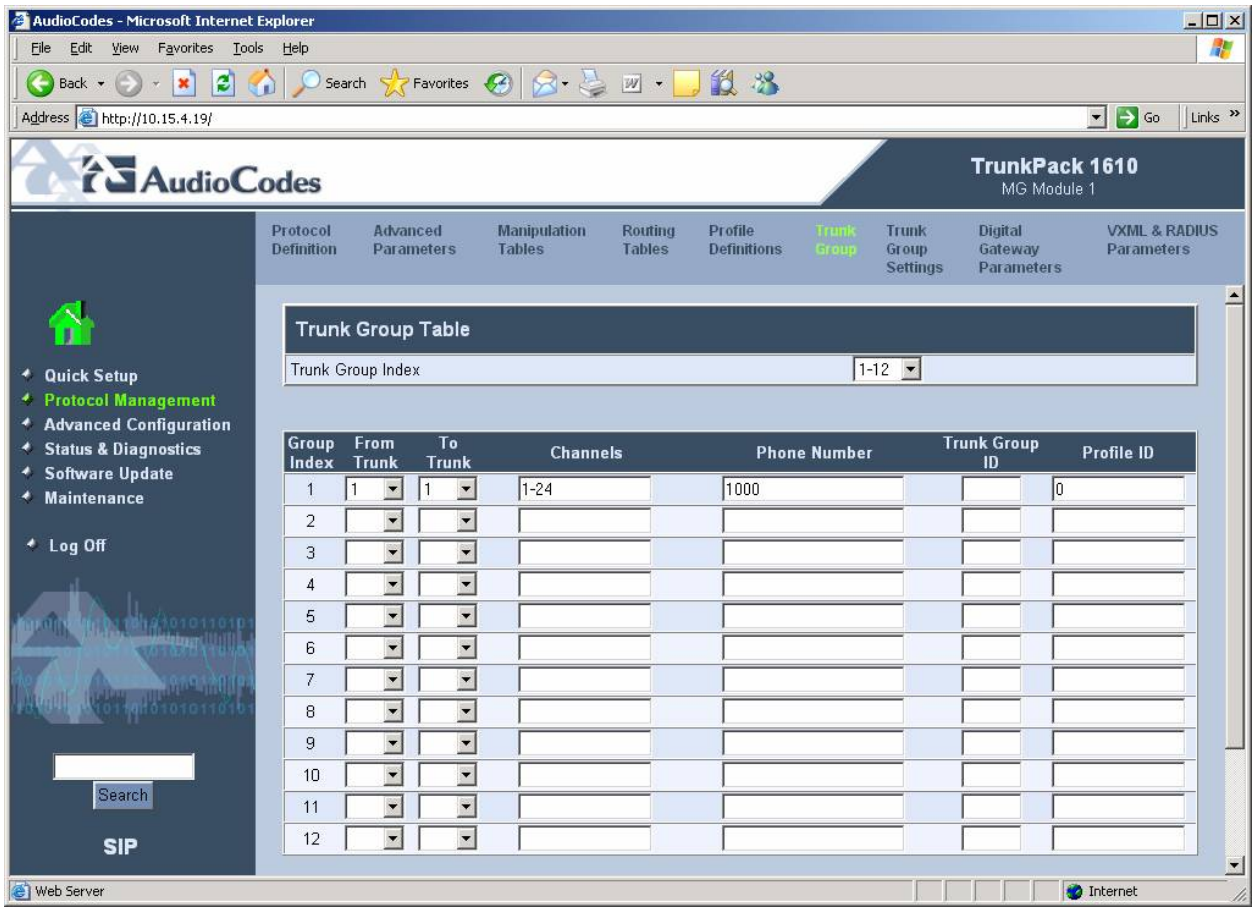


Step 6: Trunk Group Setup

There are two options to setup the Trunk Group:

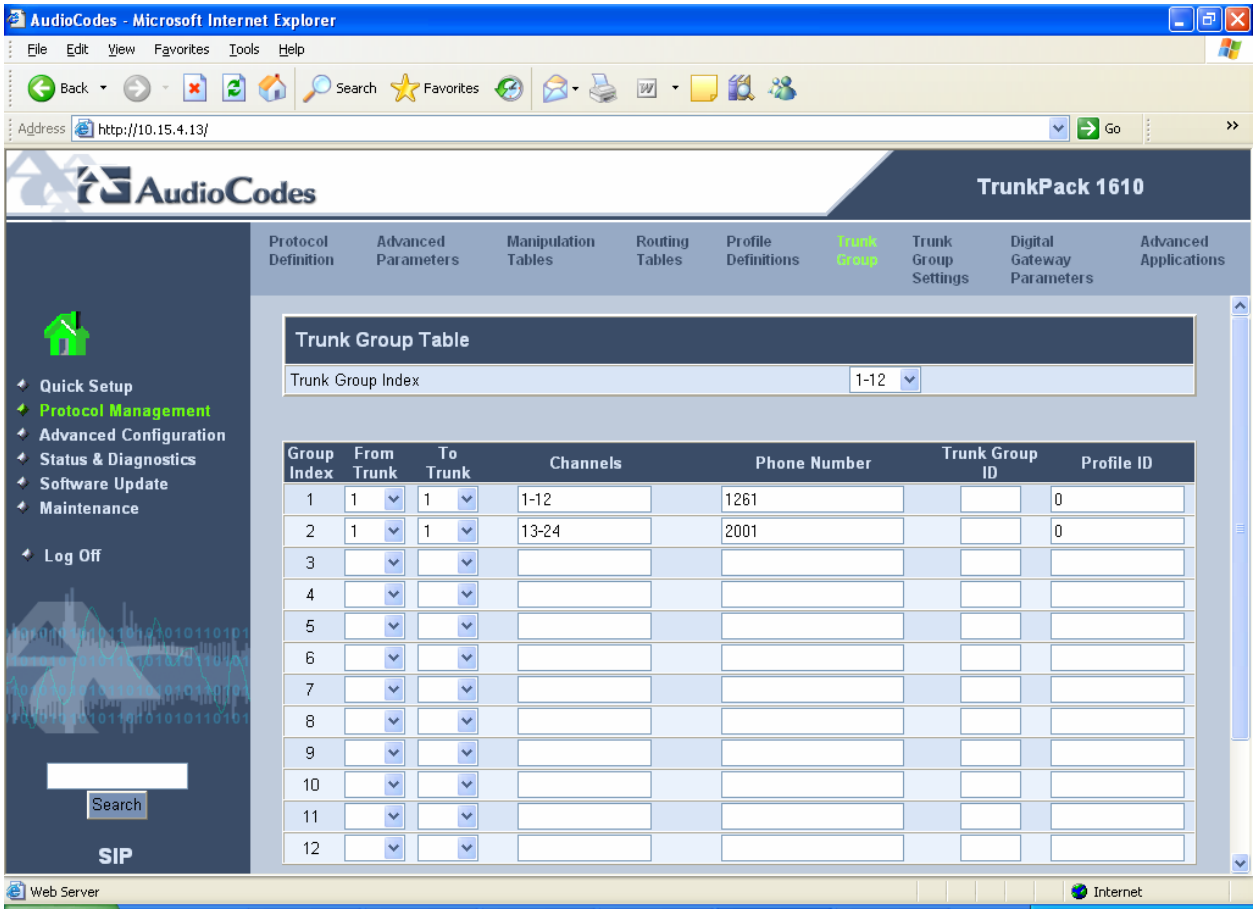
- For general deployments, use the trunk group setup as described in Step 6(a).
- For Auto Attendant and Unified Messaging combined solution, use Step 6(a) or Step 6(b), which is an example used for our specific testing environment.

Step 6 (a): Standard Trunk Group Setup

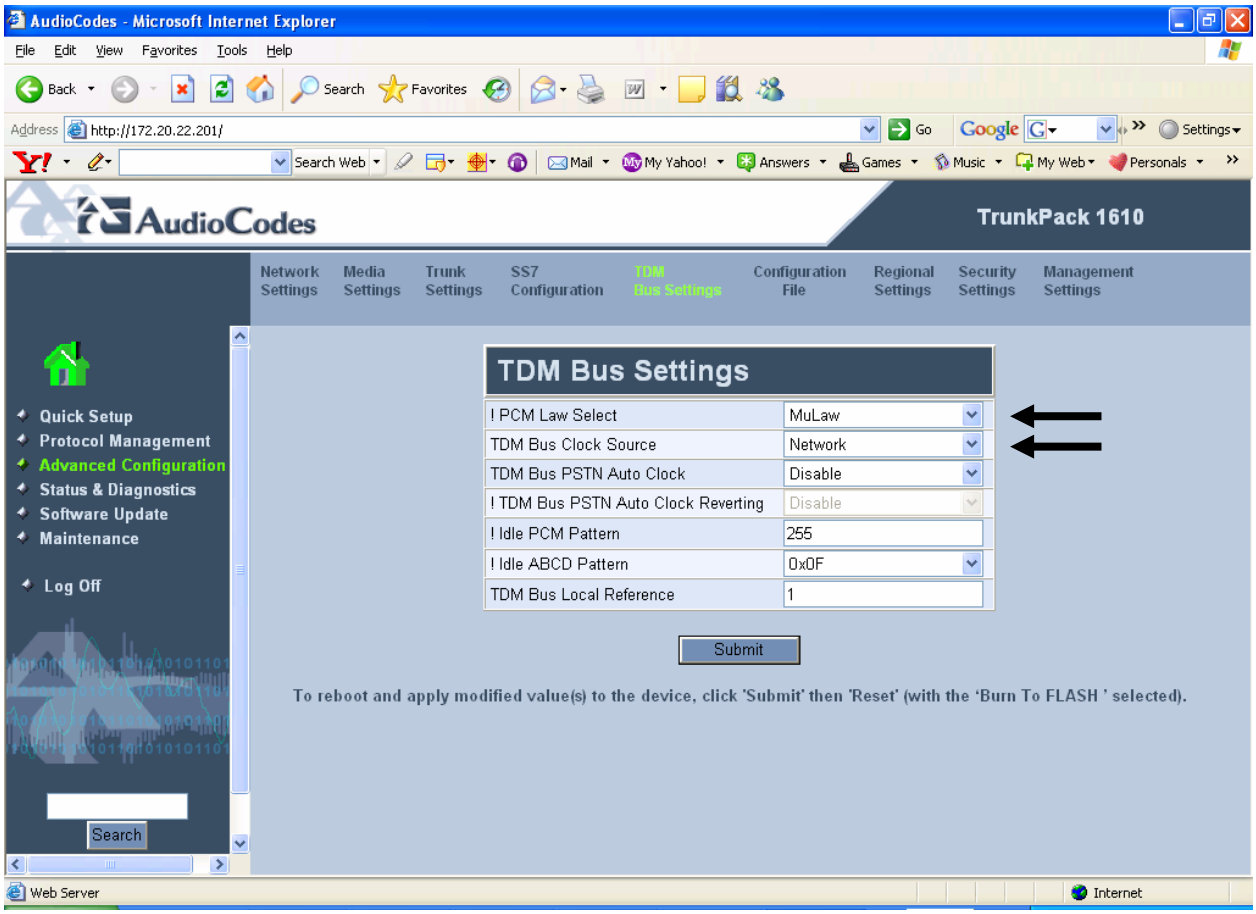


Step 6 (b): Trunk Group Setup Example

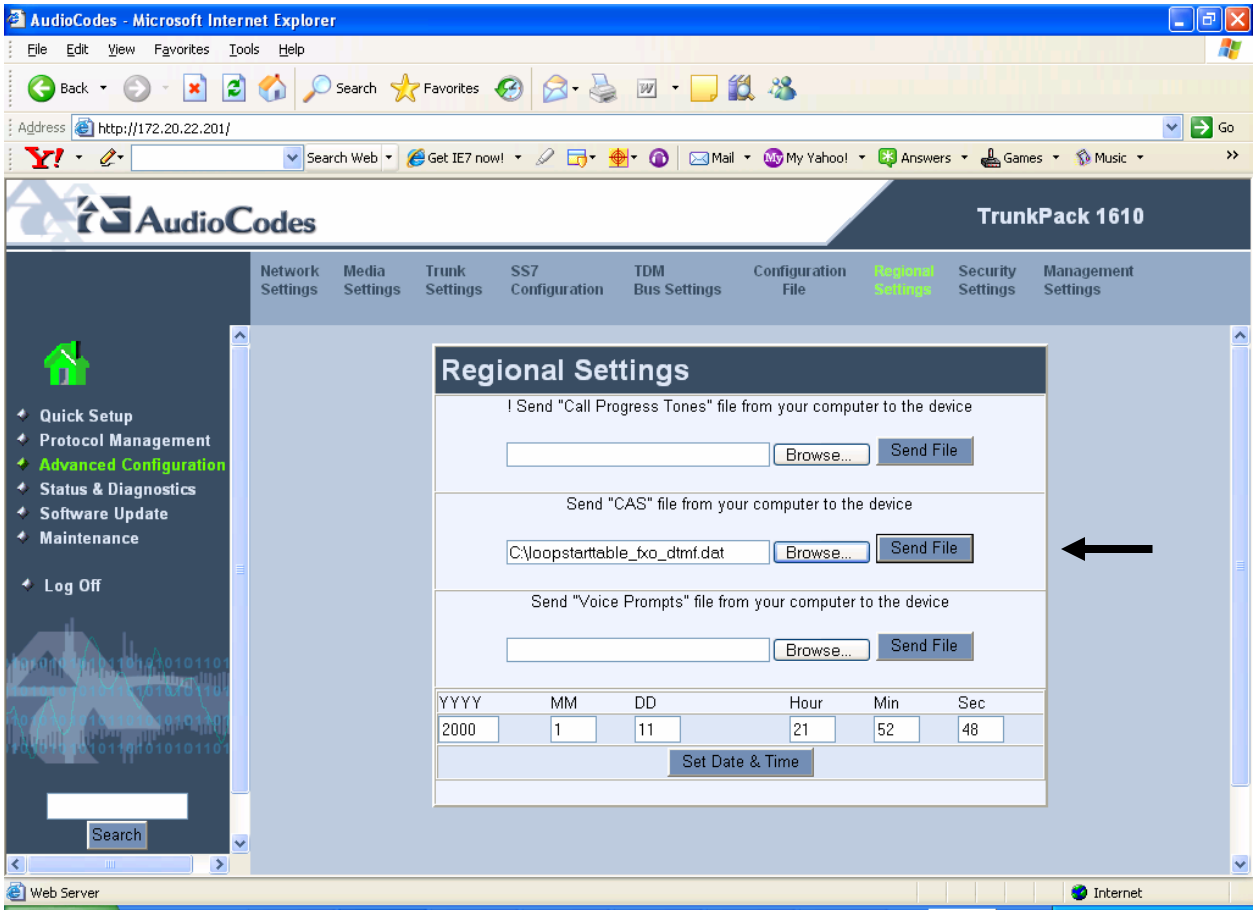
This configuration represents a split T1 configuration where the first 12 trunks of the T1 are used for Voice Mail access and the second 12 trunks are used for Auto Attendant functionality. Calls placed on the last 12 trunks of the T1 span, ring in with a phone number of *2001* through *2012* (consequently, the "20" value in Step 5 (b)). Any value can be used for the phone number as long as it doesn't conflict with Inter-Tel or UM Dial Plan.



Step 7: TDM BUS Setting

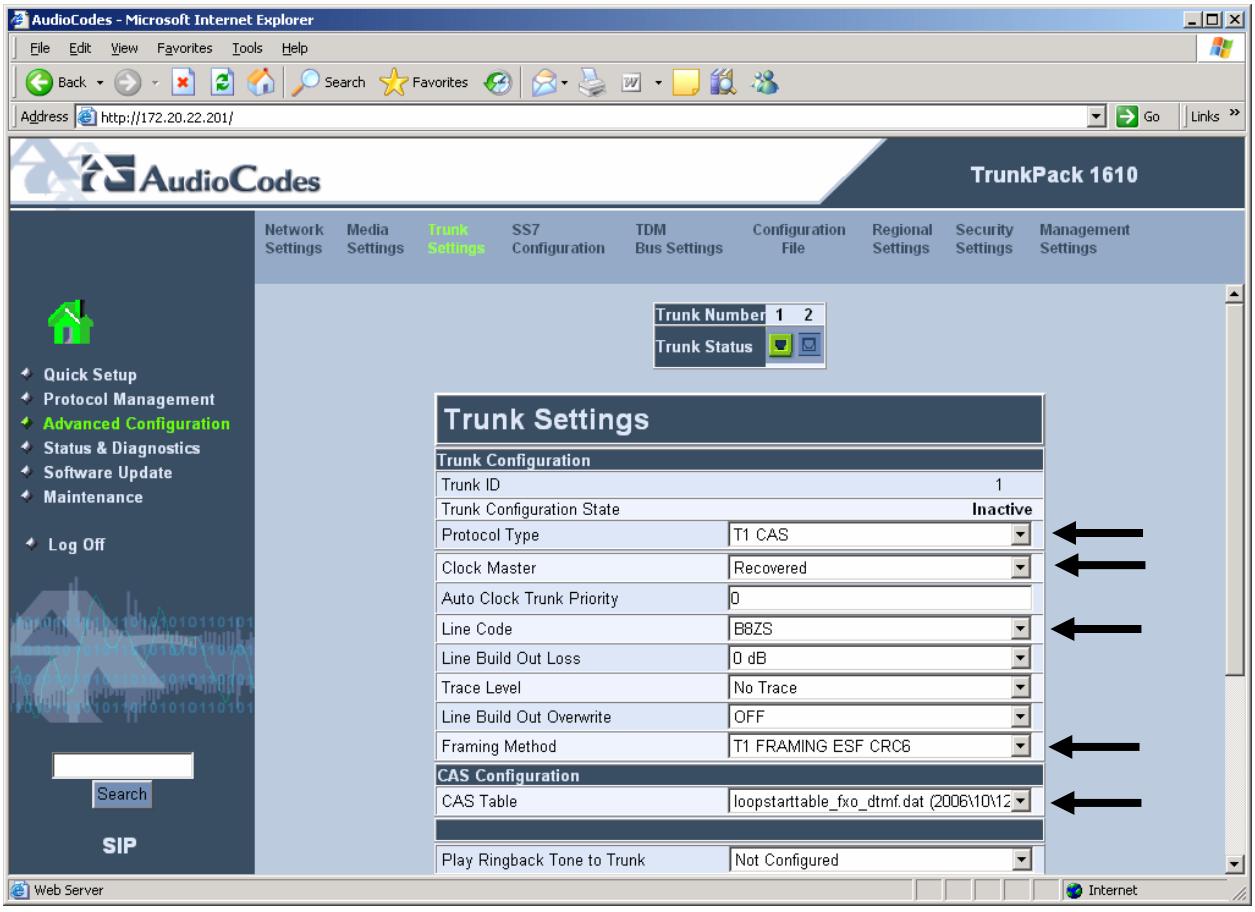


Step 8: CAS Table Download

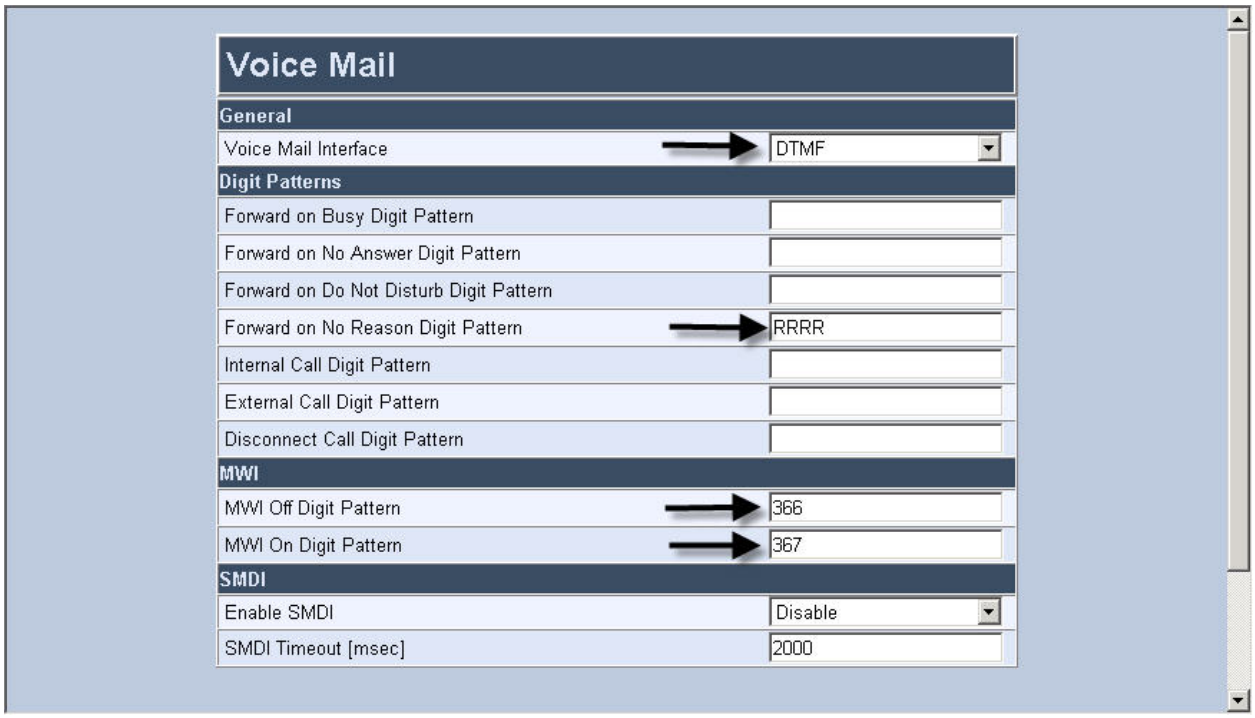


Note: The CAS file must be downloaded to the Mediant 2000 prior to the Trunk configuration. A suitable CAS file is attached to this document (refer to Section 4.1).

Step 9: Trunk Settings

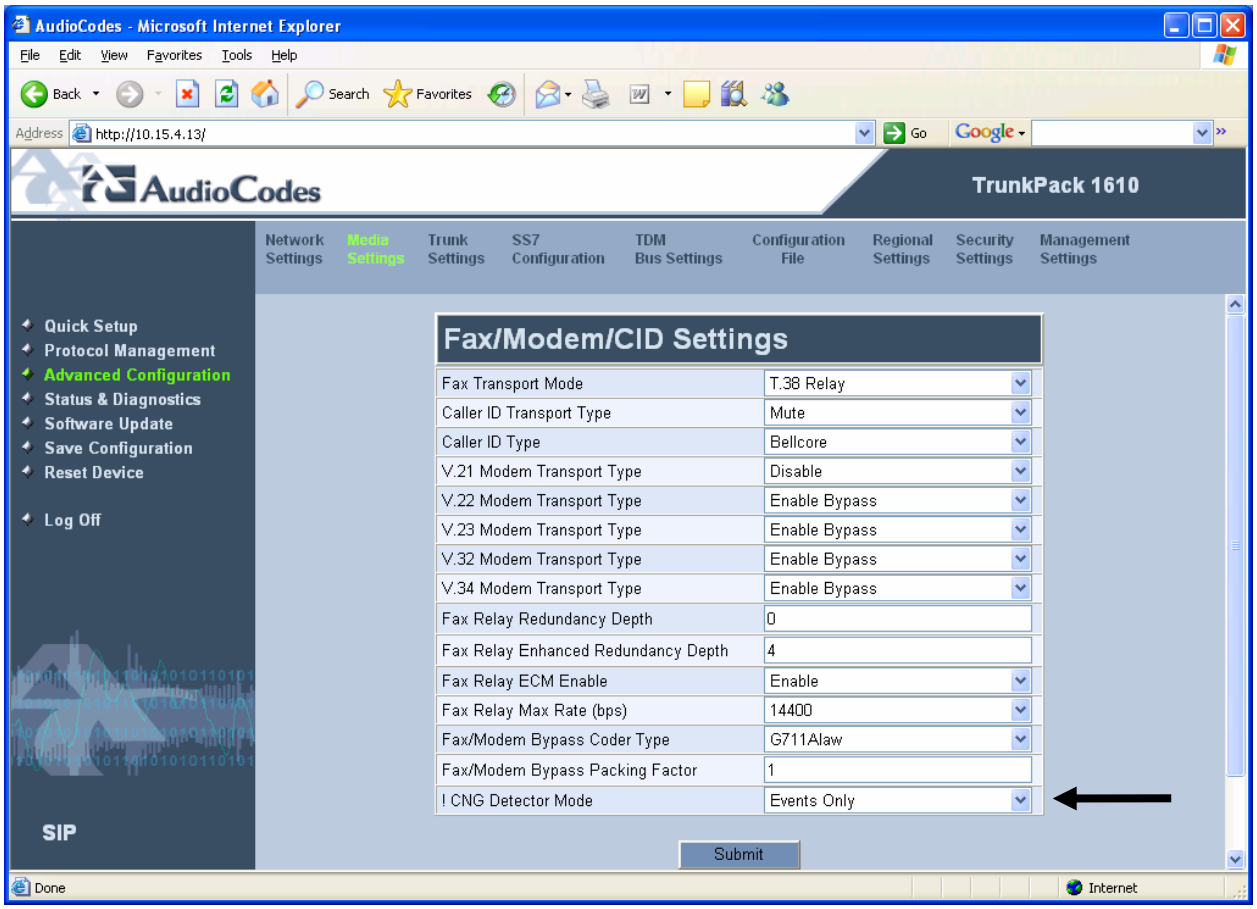


Step 10: Voicemail In-band DTMF Setup

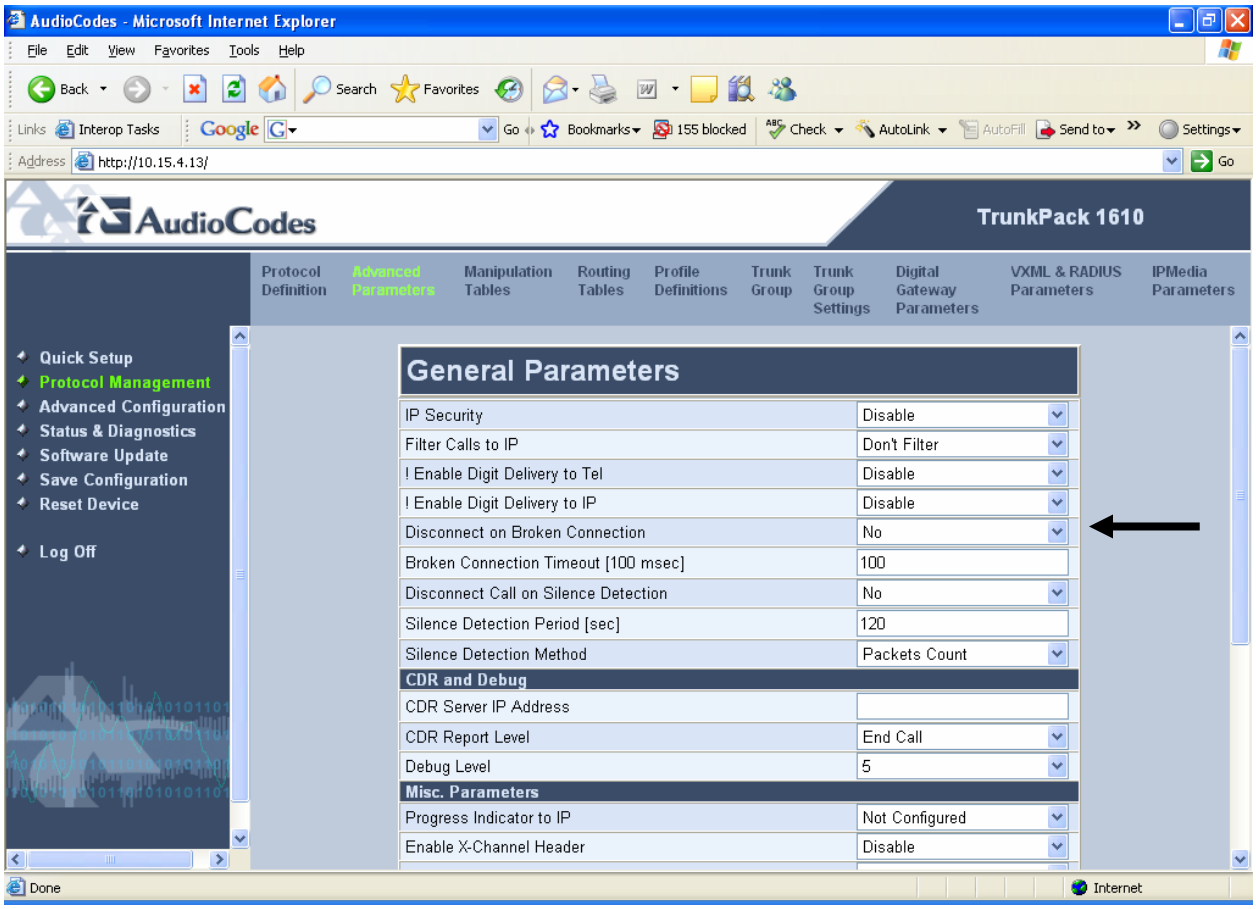


Note: The number of 'R's in the digit pattern is based on the Inter-Tel dial plan. If the system is configured with three-digit extensions, then only three 'R's are necessary, etc. The MWI patterns are based on the system default feature codes. Use the appropriate feature codes for the system being configured.

Step 11: FAX Setup



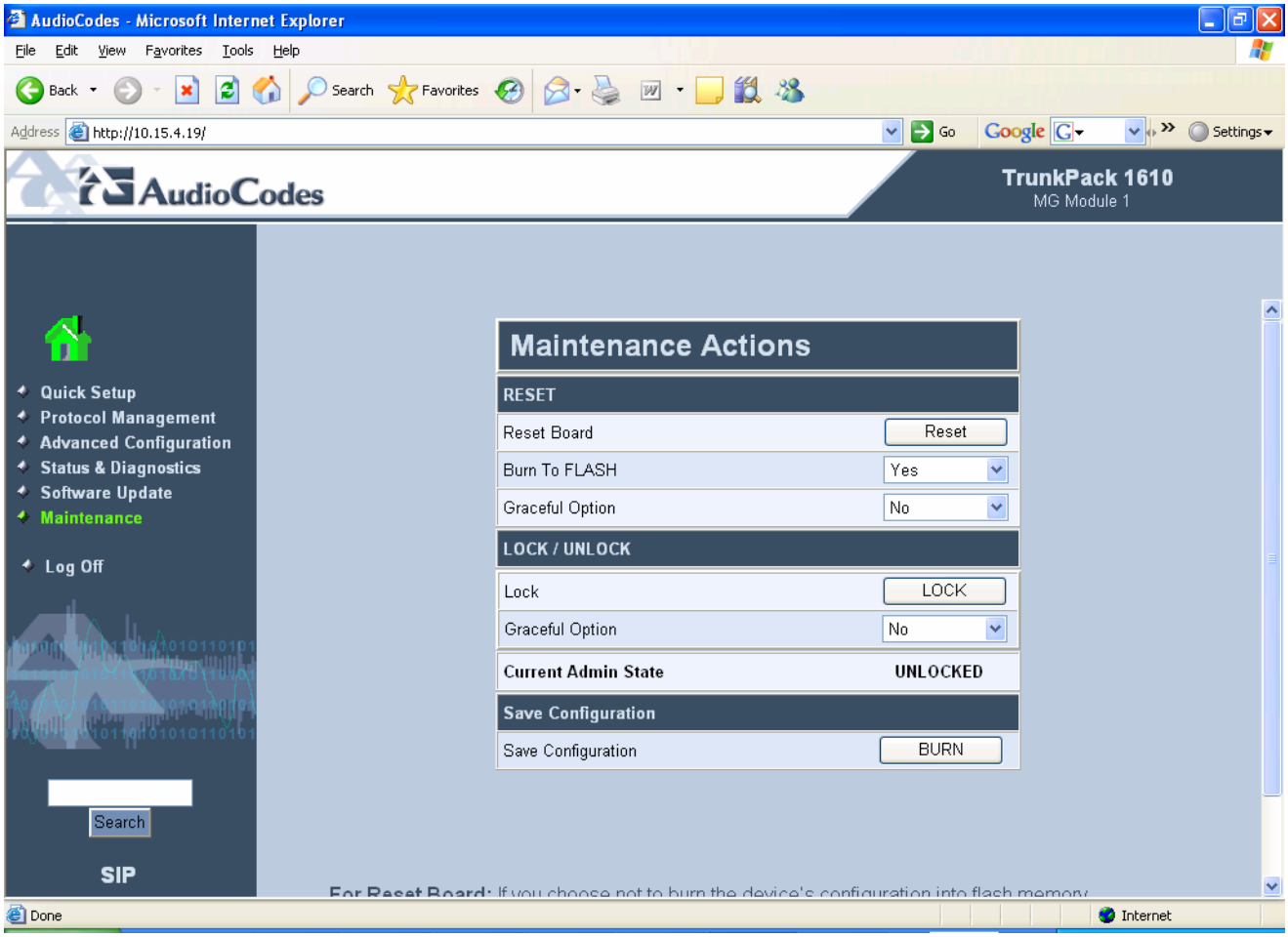
Step 12 : General Setup



Step 13: INI file

- EnableMWI = 1
- SubscriptionMode = 1
- IsSpecialDigits = 1
- LineTransferMode = 1
- EnableDetectRemoteMACChange = 2
- ECNLPMode = 1
- TrunkTransferMode_X = 3 (where 'X' represents the Trunk number based on a 0 index, e.g., for Trunk 1, 'X' = 0)

Step 14: Reset Mediant 2000



Click **Reset** to reset the gateway.

4.1. Configuration Files

The ZIP file includes the following files:

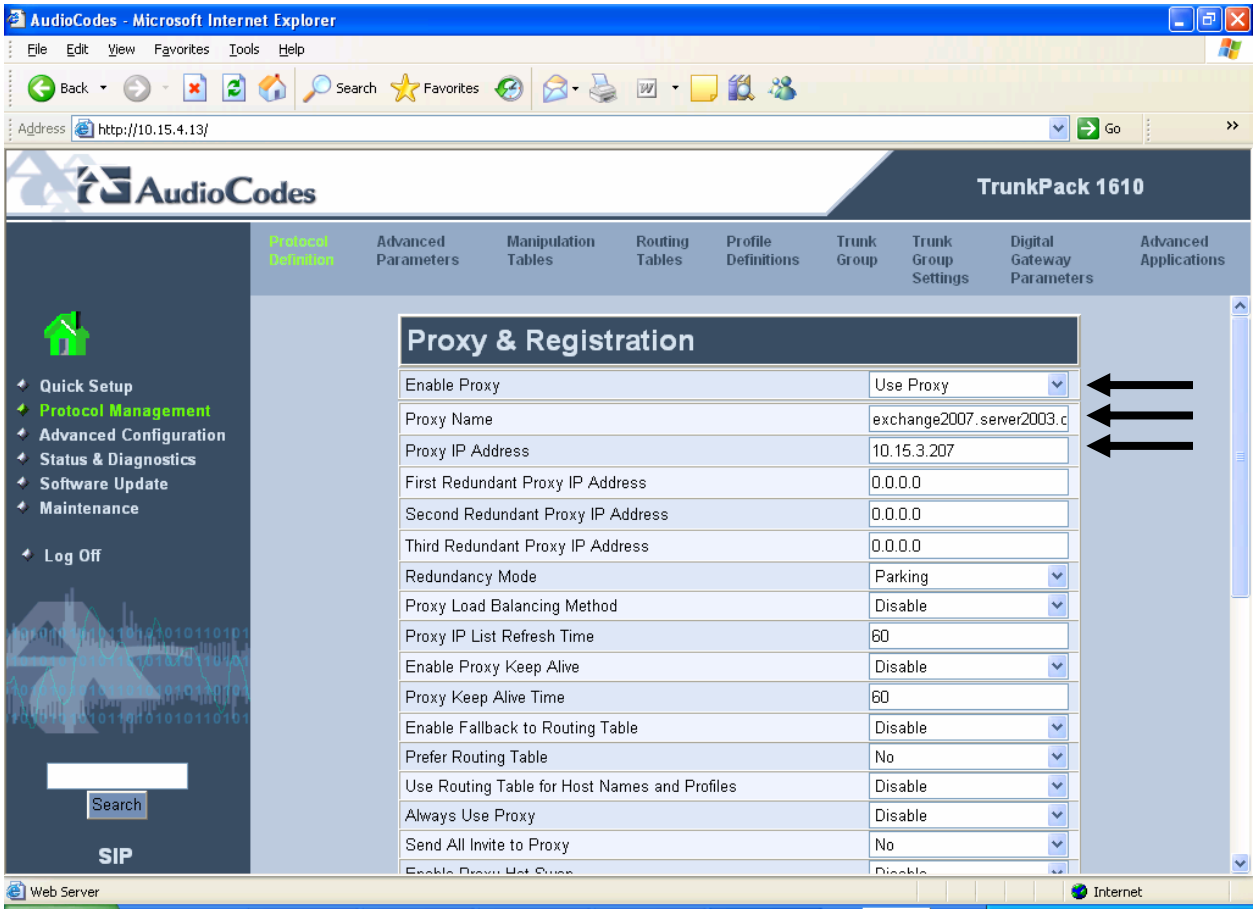
- Audiocodes configuration ini file (.ini file extension).
- Audiocodes T1 CAS file (.dat file extension).

AudioCodes Files - Inter-Tel 5000.zip

4.2. TLS Setup

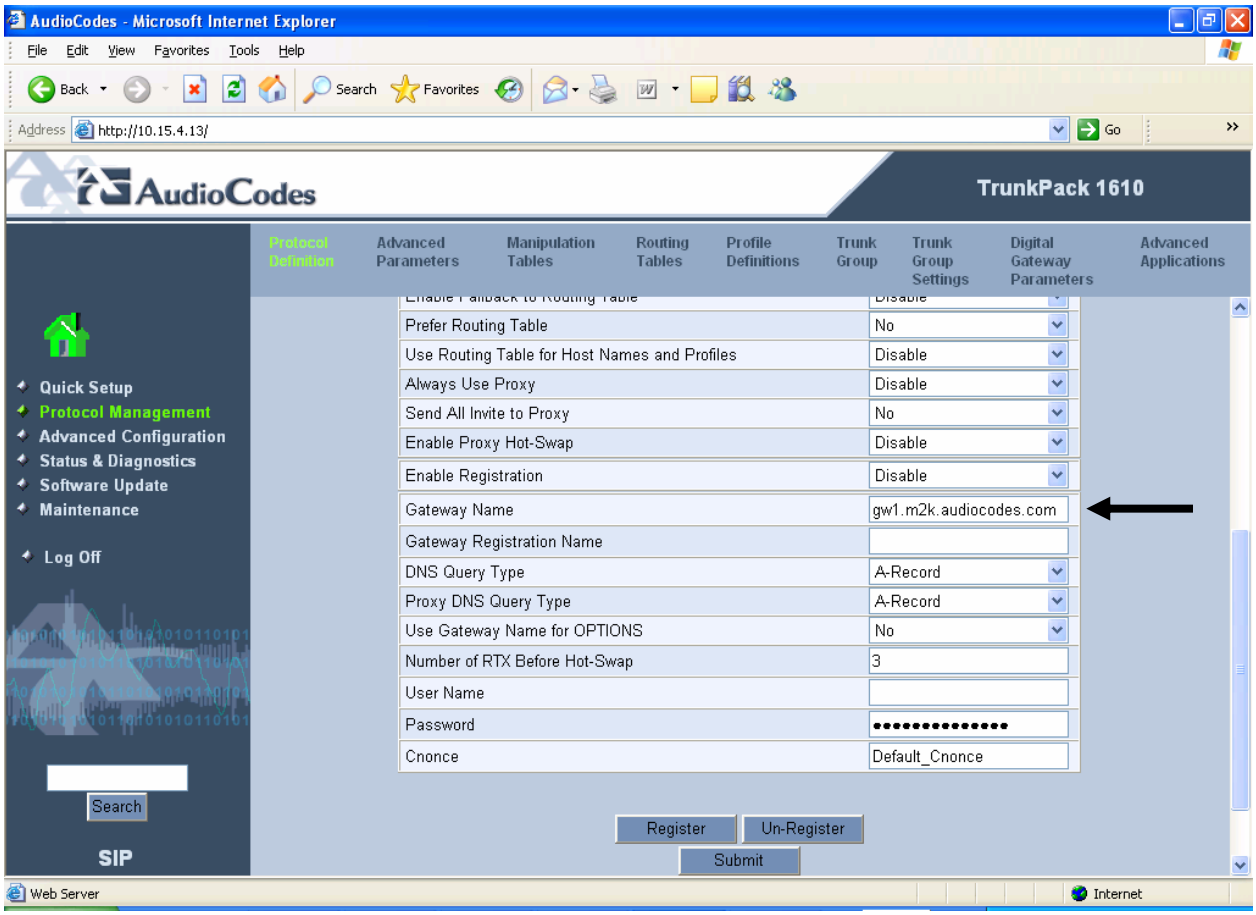
Did not perform TLS setup.

Step 1: PBX to IP Routing Setup



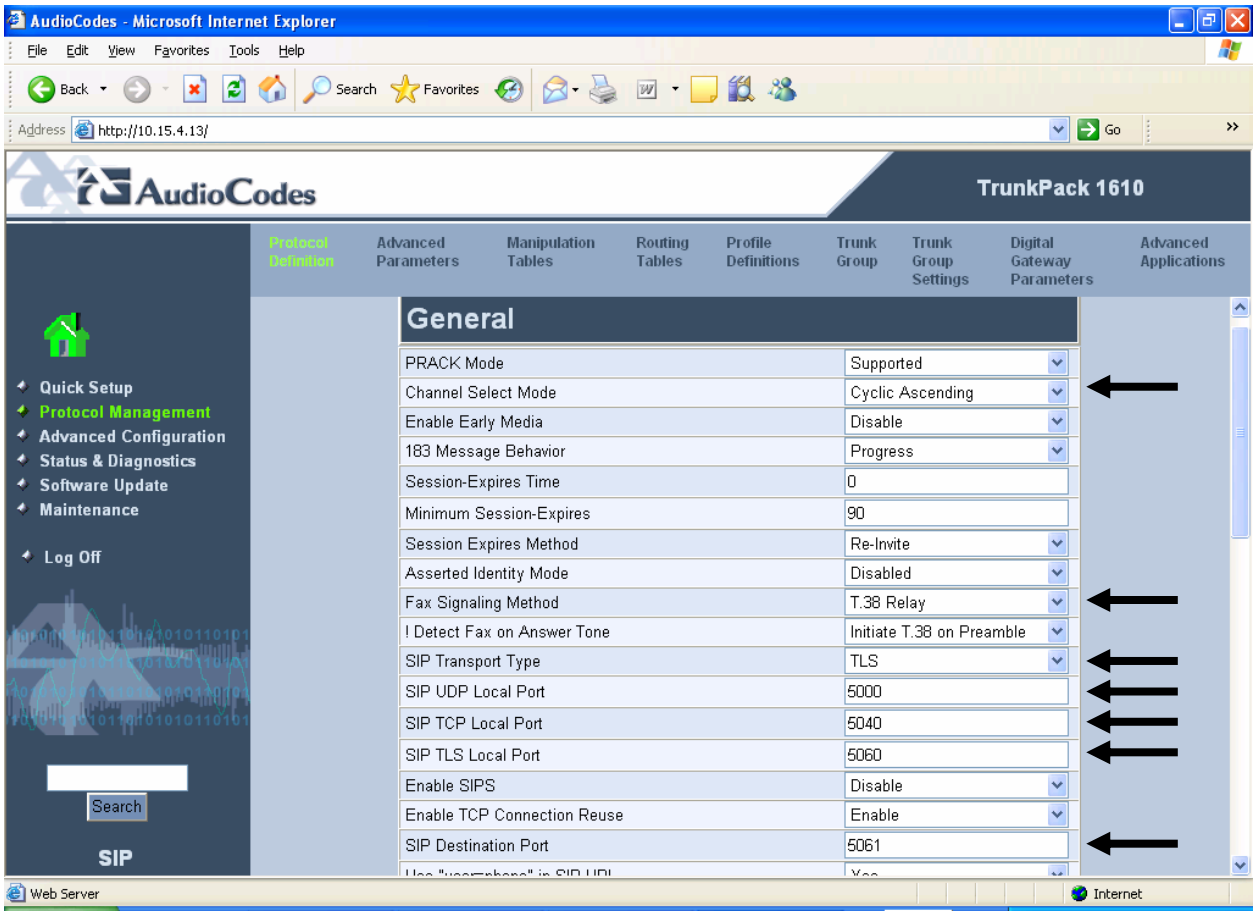
Note: The Proxy IP Address and Name must correspond to the network environment in which the Microsoft Unified Messaging server is installed (for example, 10.15.3.207 for IP Address and exchaneg2007.server2003.com for the FQDN of the Microsoft Unified Messaging host).

Step 2: SIP Environment and Gateway Name Setup

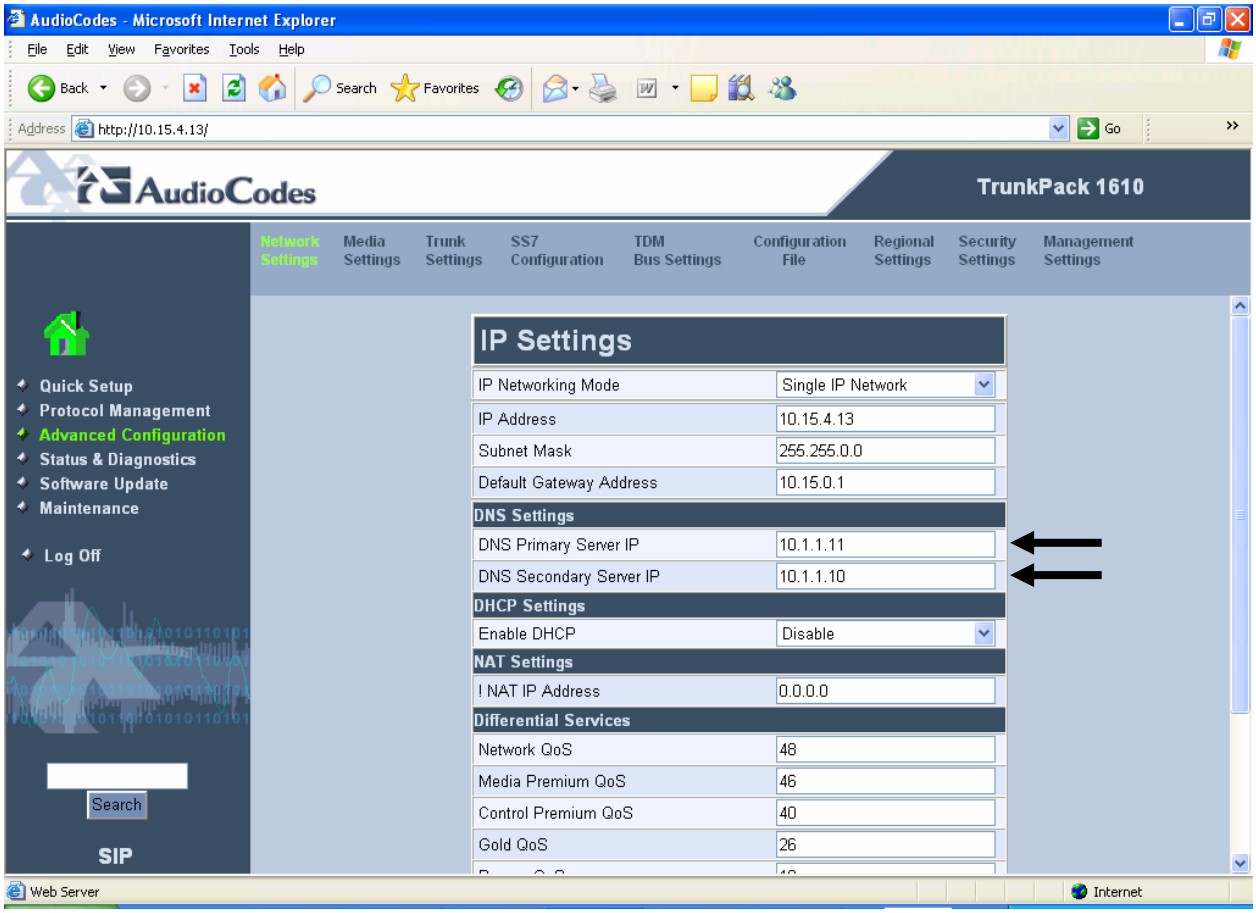


Note: Assign an FQDN name to the gateway (for example, gw1.m2k.audiocodes.com). Any gateway name that corresponds to your network environment is applicable; the only limitation is to exclude underscores (_) in the name (Windows Certification server limitation).

Step 3: SIP Environment Setup (Cont.)

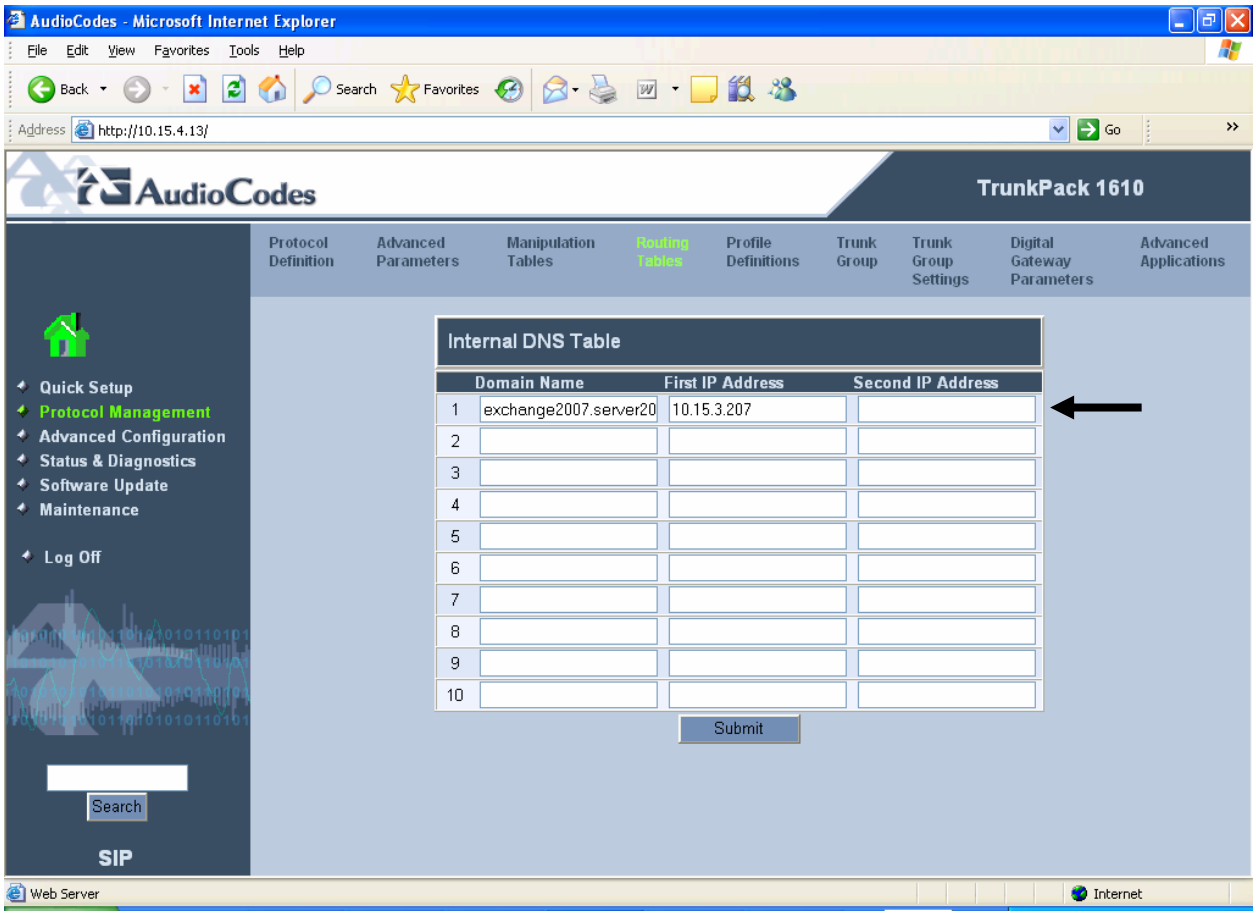


Step 4: DNS Servers Setup



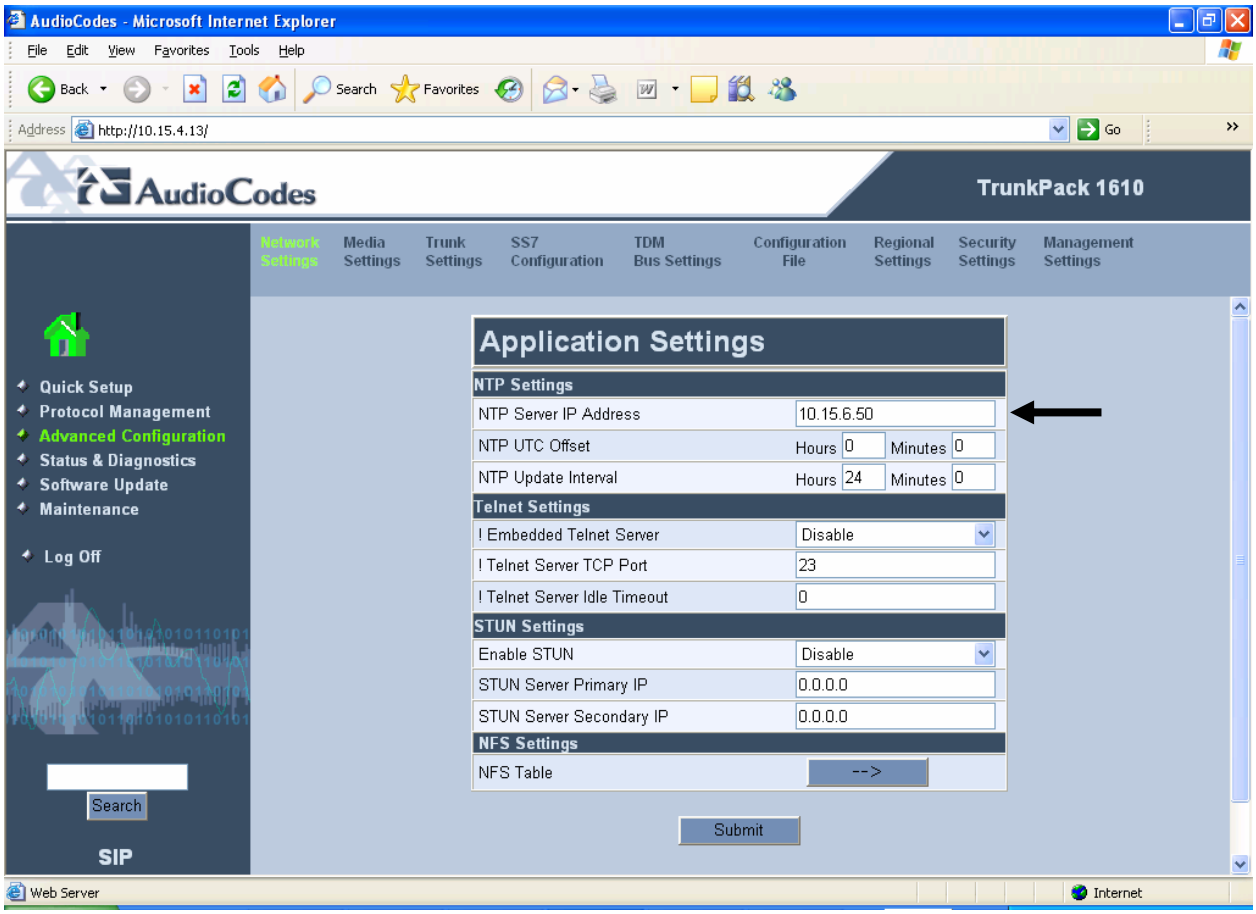
Note: Define the primary and secondary DNS servers' IP addresses so that they correspond to your network environment (for example, 10.1.1.11 and 10.1.1.10). If no DNS server is available in the network, then skip this step.

Step 5: Internal DNS Setup



Note: If no DNS server is available in the network, define the internal DNS table where the domain name is the FQDN of the Microsoft Unified Messaging server and the First IP Address corresponds to its IP address (for example, exchange2007.com and 10.15.3.207).

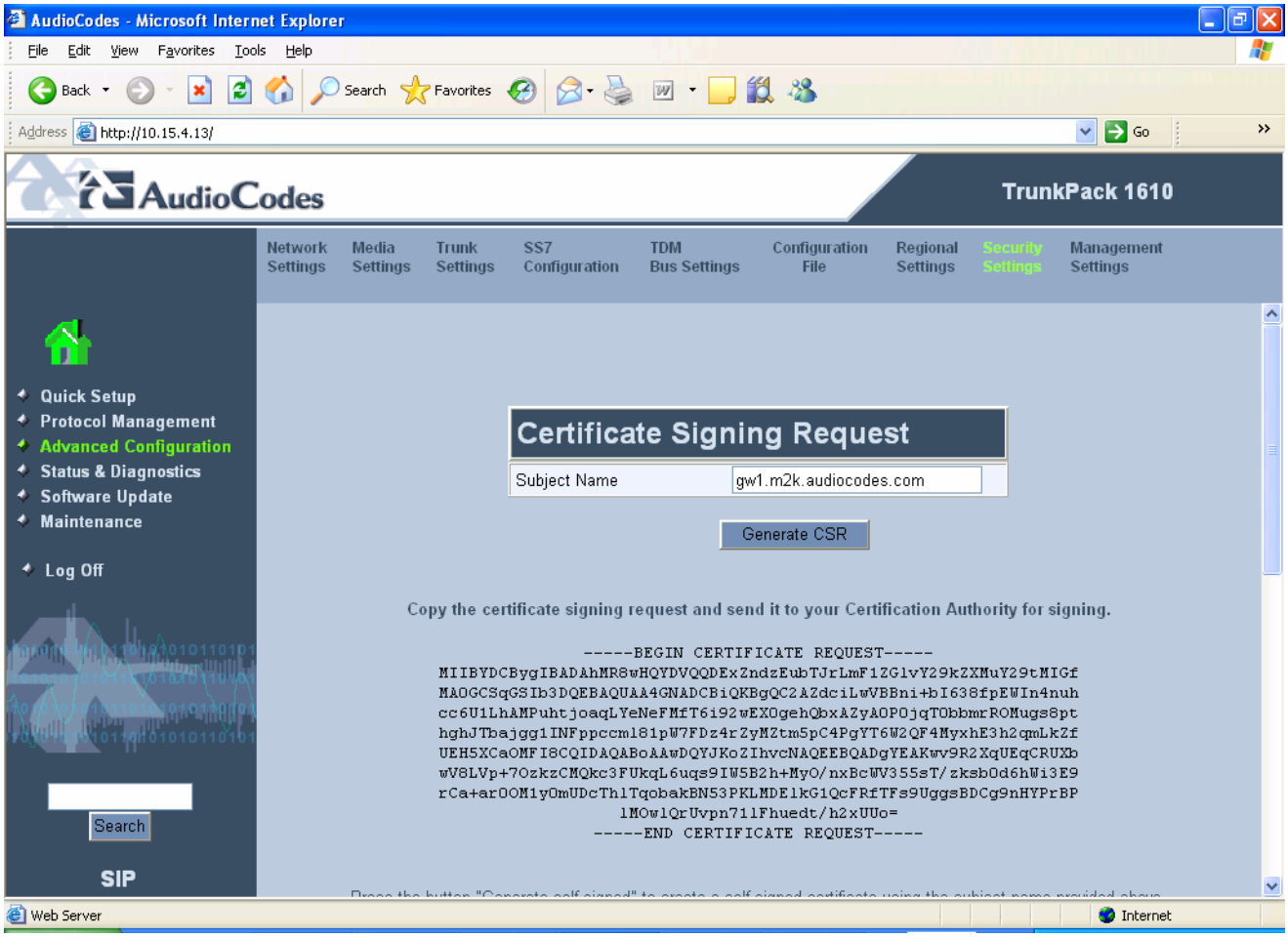
Step 6: NTP Server Setup



Note: Define the NTP server's IP address so that it corresponds to your network environment (for example, 10.15.3.50). If no NTP server is available in the network, then skip this step (as the gateway uses its internal clock).

Step 7: Generate Certificate Setup

Use the screen below to generate CSR. Copy the certificate signing request and send it to your Certification Authority for signing.

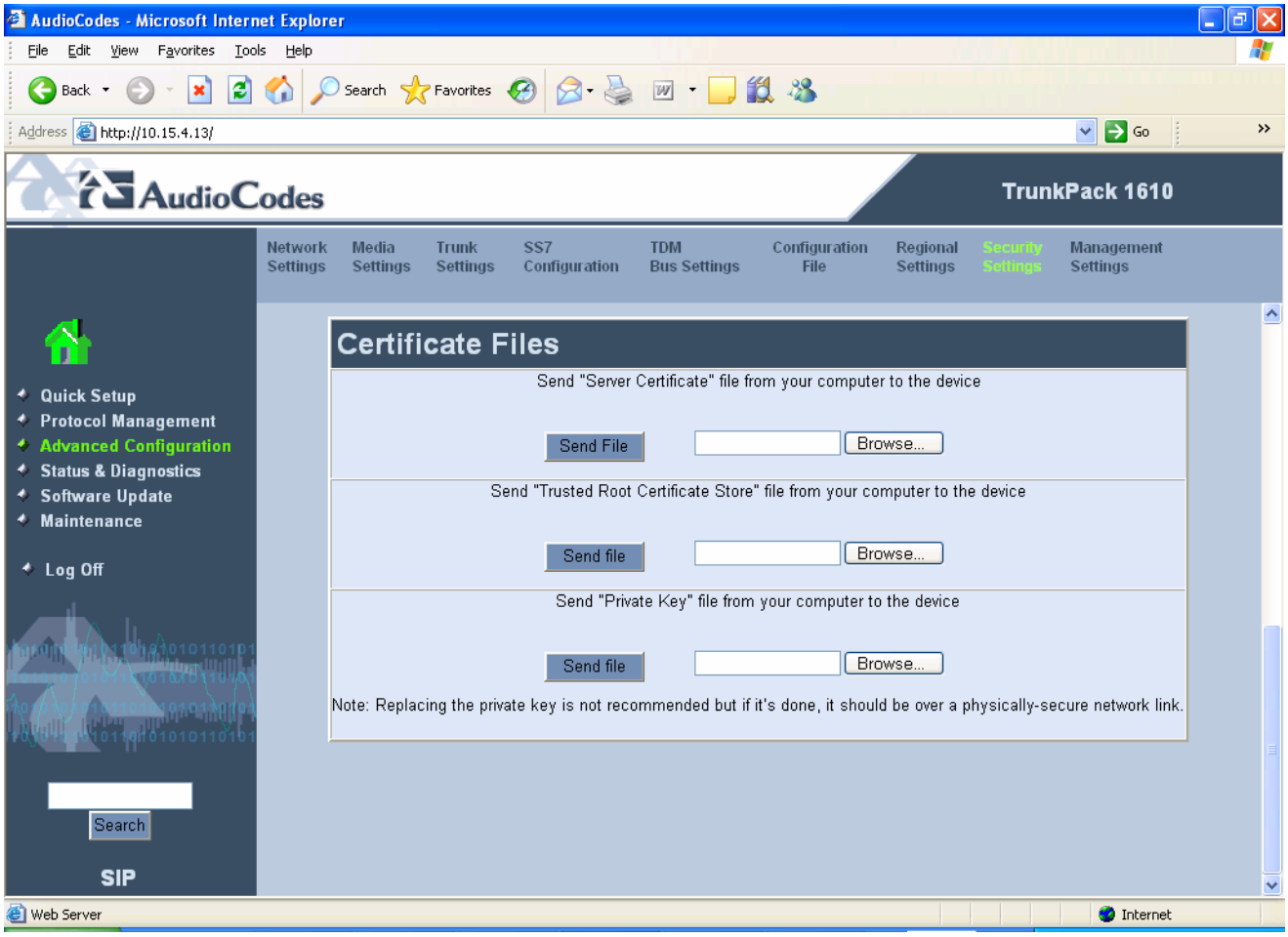


Step 8: Uploading Certificates Setup

The screen below is used to upload the sign certificates.

In the "Server Certificate" area, upload the gateway certificate signed by the CA.

In the "Trusted Root Certificate Store" area, upload the CA certificate.



5. PBX Setup Notes

The configuration procedures described in this document assume that the system administrator is well versed in the Inter-Tel 5000 Database Programming Interface. In addition, this description assumes that all mailboxes have been enabled on the Exchange 2007 UM Server and have been enabled for Unified Messaging. These instructions don't describe how to configure Exchange 2007.

Information Gathering

The configuration of the Inter-Tel 5000 system is independent of the extensions used on the Exchange 2007 Server for Auto Attendant or Subscriber Access. It's only necessary to know the Auto Attendant extension for the Audiocodes gateway configuration.

It's necessary to know which gateway is being used and how many ports on the gateway are to be dedicated to Exchange 2007 UM integration. These instructions assume a single T1 interface on the Mediant series gateways, and four FXS ports on the MediaPack series gateways. The instructions are easily scaleable to include additional ports as required.

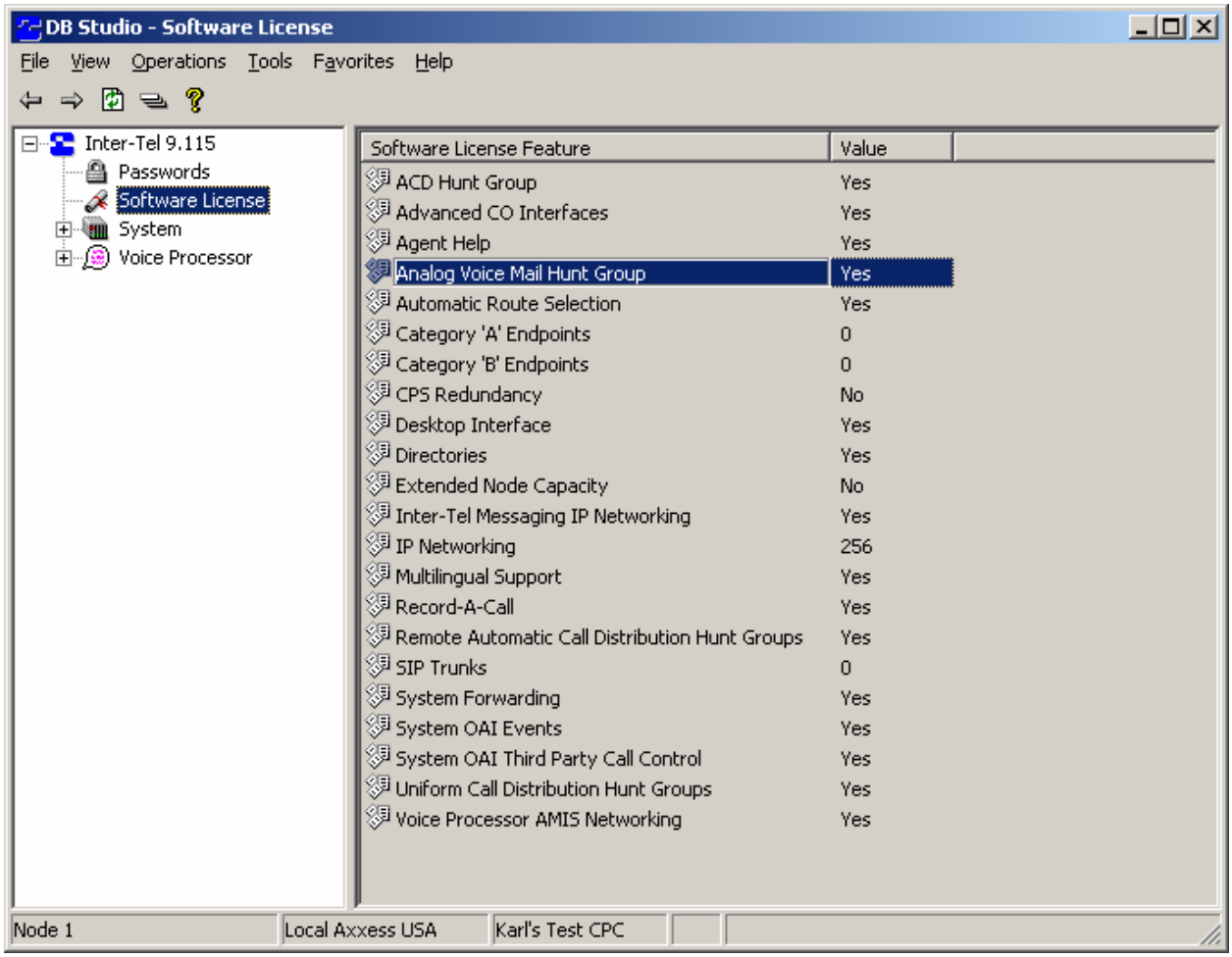
Finally, these instructions assume that the first 12 circuits on the T1 are used for voice mail access and the last 12 are used for Auto Attendant functionality. Similar for the FXS ports: two for voice mail and two for Auto Attendant.

System Limitations

- The Inter-Tel 5000 doesn't support the sending of the calling user number for direct calls (i.e., user calls UM to retrieve voice message). Therefore, when the user dials directly to the Exchange 2007 UM Server, the user hears the general welcome prompt: "Welcome, you are connected to Microsoft Exchange, to access your mailbox, enter your extension.", at which the user is required to enter the user's extension number in addition to the pin number.
- Station Speed Dials (which would expedite this process) doesn't function. The system doesn't allow the passing of digits beyond the completed extension on a hunt group call. For example, if one were to program a station speed dial string of "1260P1200" where 1260 is the voice mail hunt group, then the "P1200" is ignored.
- Since the integration is over T1-CAS, there is no Calling Party Number or Caller ID that is passed to the Exchange server. Therefore, all the voicemails appear to have been left by the Analog Voice Mail Hunt Group pilot number.
- Microsoft Exchange 2007 doesn't support Message Waiting Indications (MWI). The gateway instructions indicates where to setup the "on" and "off" MWI digit patterns in the event that this functionality is added in a later release of Exchange.

Software License Features

The "Analog Voice Mail Hunt Group" feature needs to be enabled on the software license.



System Configuration

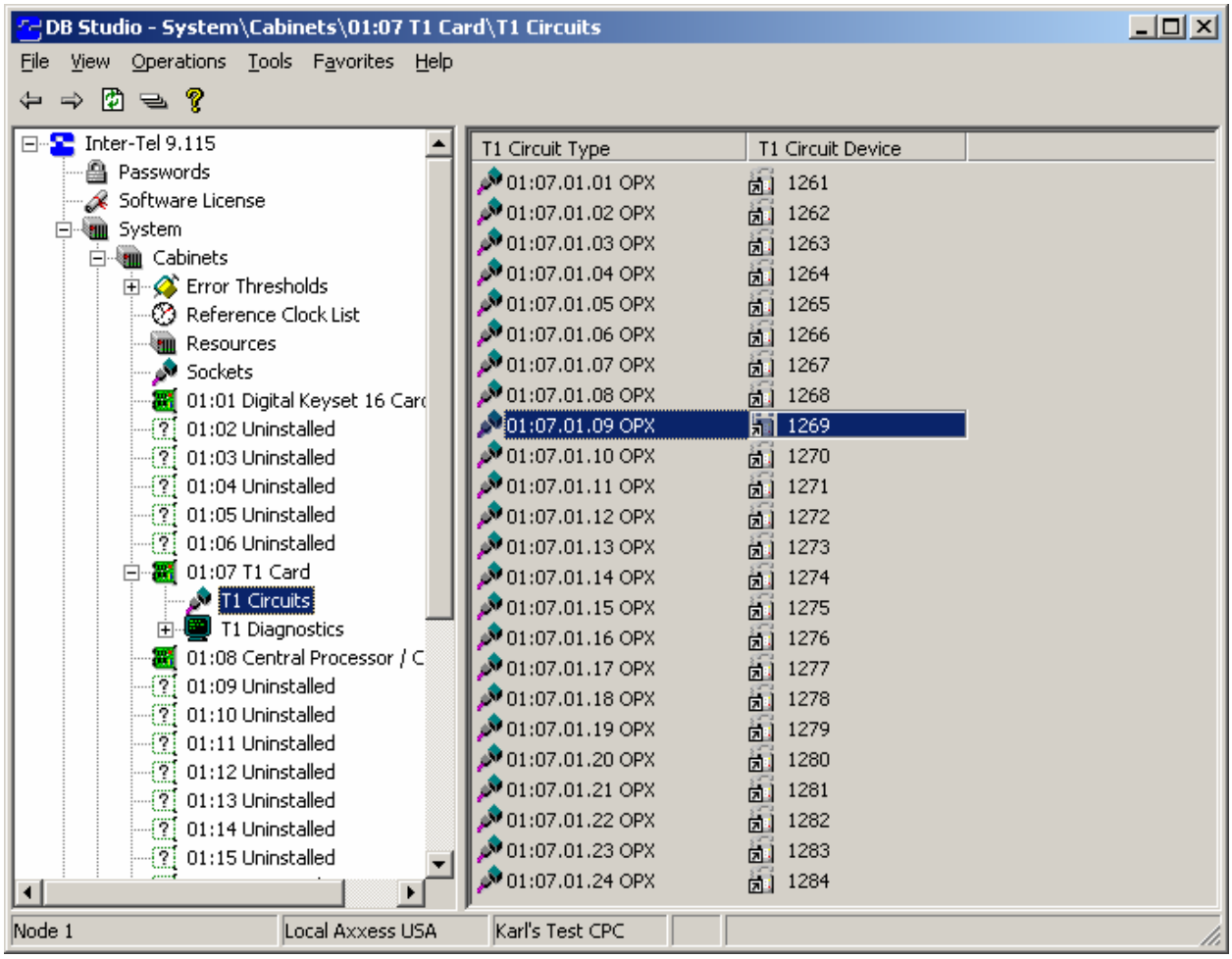
Step 1: INI file Inter-Tel 5000 Communication Server Configuration

There are two options when it comes to implementing Exchange 2007 UM. Both involve the configuration of Single Lines to interface with the Audiocodes gateway.

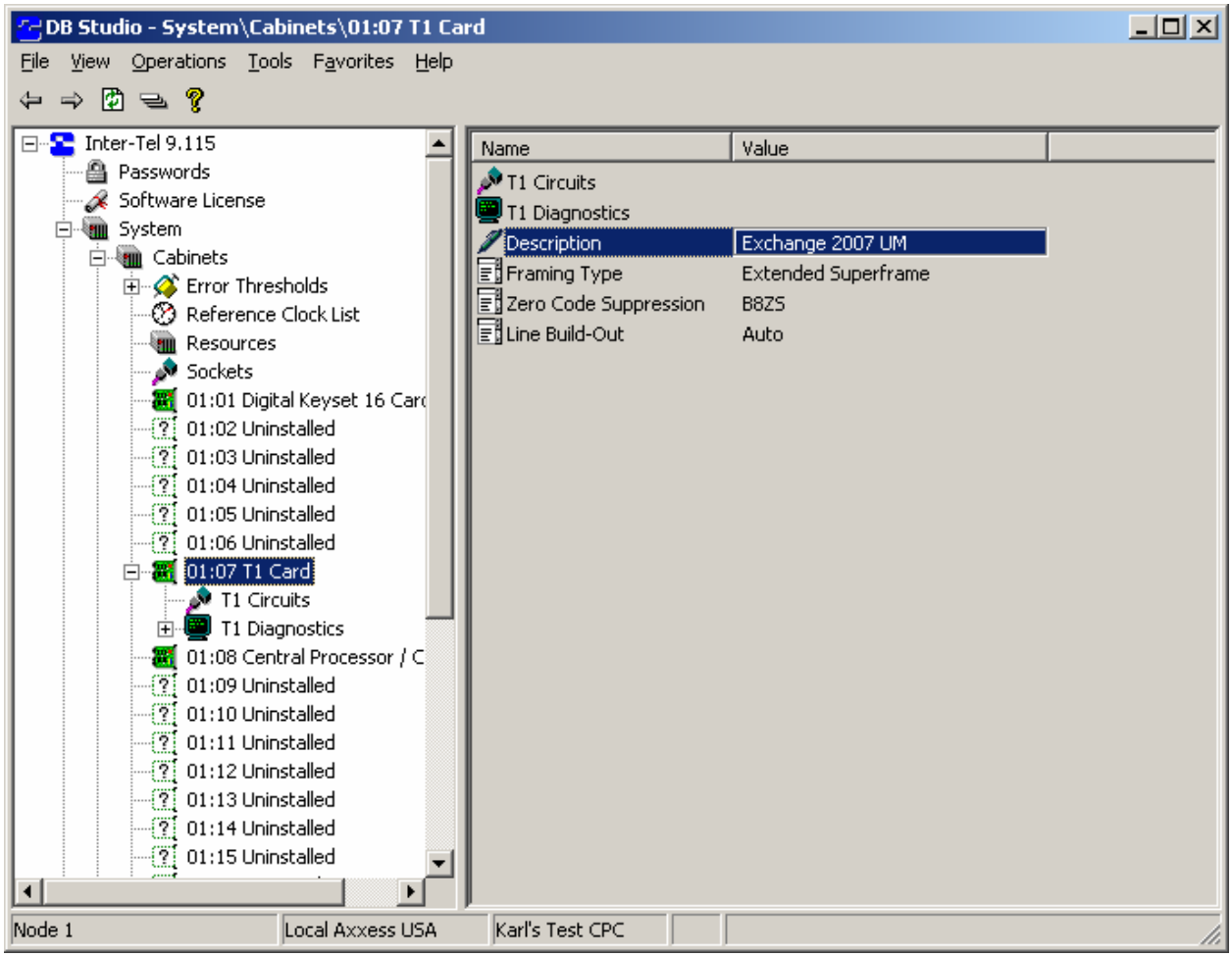
- For the Mediant series gateways, one of the T1 bays needs to be configured for a T1 Module. If the plan is to use more than one T1 span on the gateway, then additional bays may need to be configured.
- For the MediaPack series gateways, one of the bays needs to be configured for a Single Line Module. Depending on how many FXS ports are available on the deployed MediaPack, additional Single Line Modules may be required. Single Line Modules on the DEI units may be used as well.

Step 2: T1 Configuration

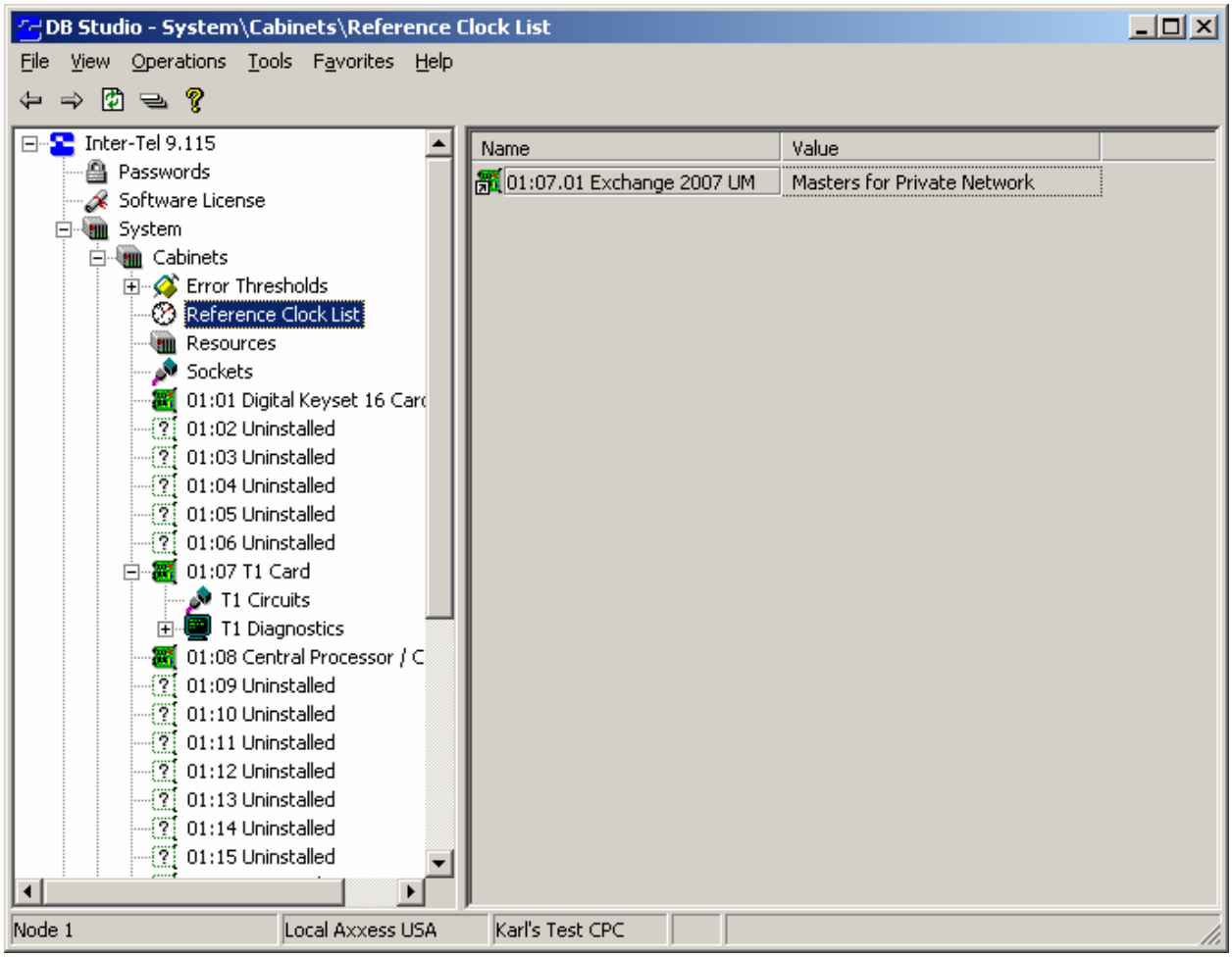
All the T1 circuits should be configured for the OPX circuit type.



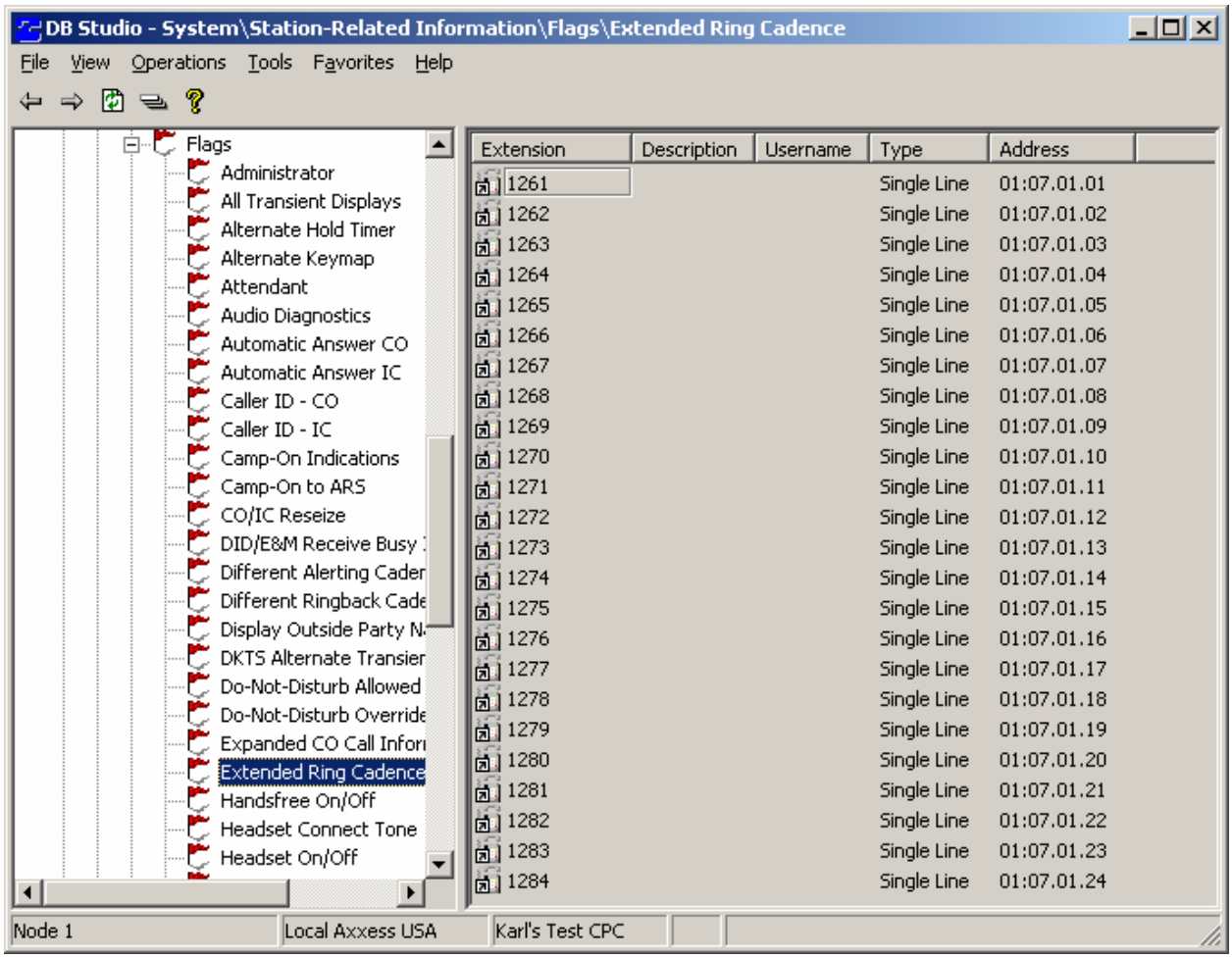
The Framing Type should be set to Extended Superframe and Zero Code Suppression should be set to B8ZS.



The T1 should be set as “Masters to Private Network” in the Reference Clock List under the System folder.



The Extended Ring Cadence flag needs to be enabled for all the OPX circuits.



Step 3: Single Line Card Configuration

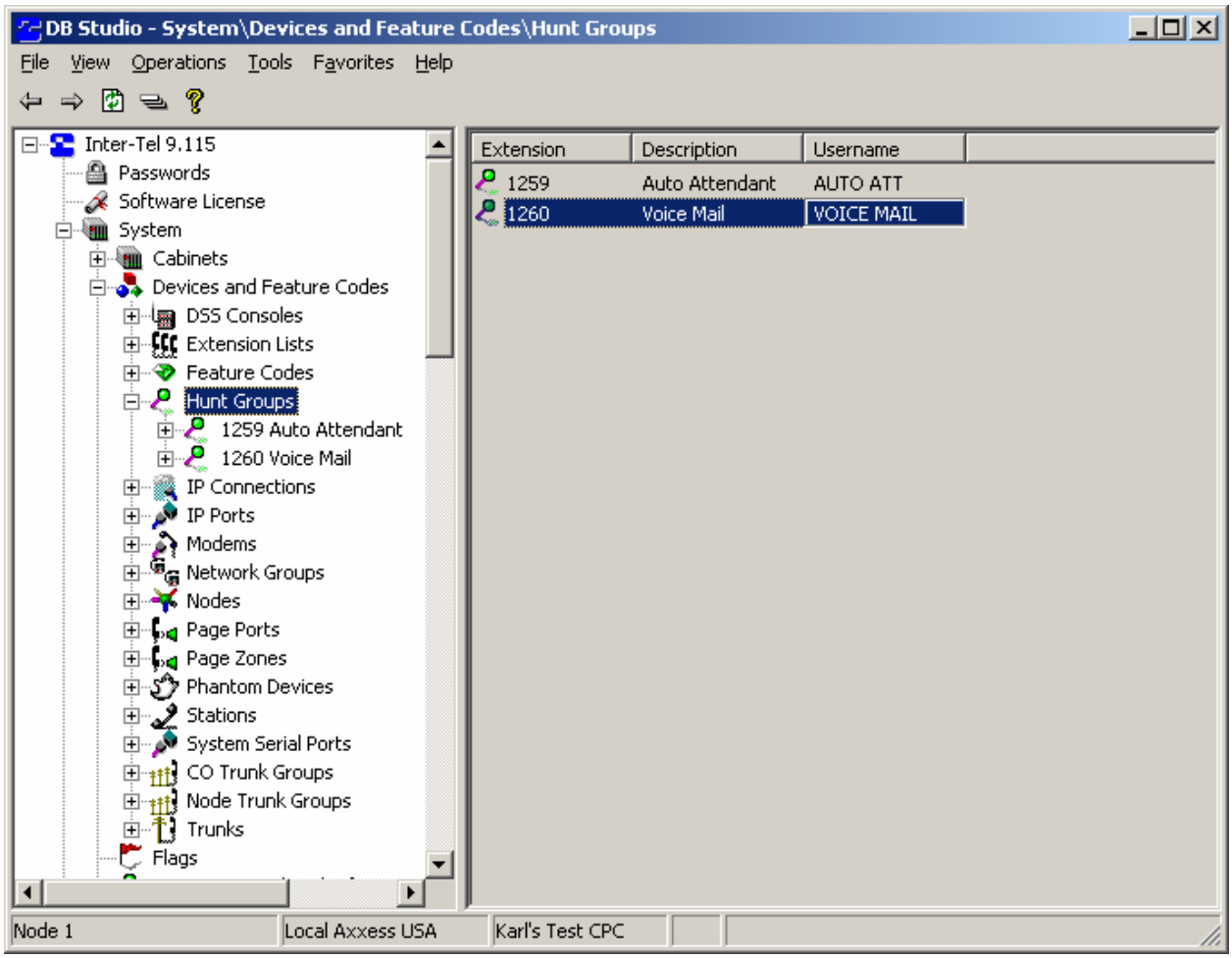
All the Single Line Module circuits should be configured for the Single Line circuit type.

The Extended Ring Cadence flag needs to be enabled for all the Single Line circuits.

Device Configuration

Step 4: Hunt Group Configuration

Two hunt groups are required: one for voice mail access and the other for auto attendant functionality.

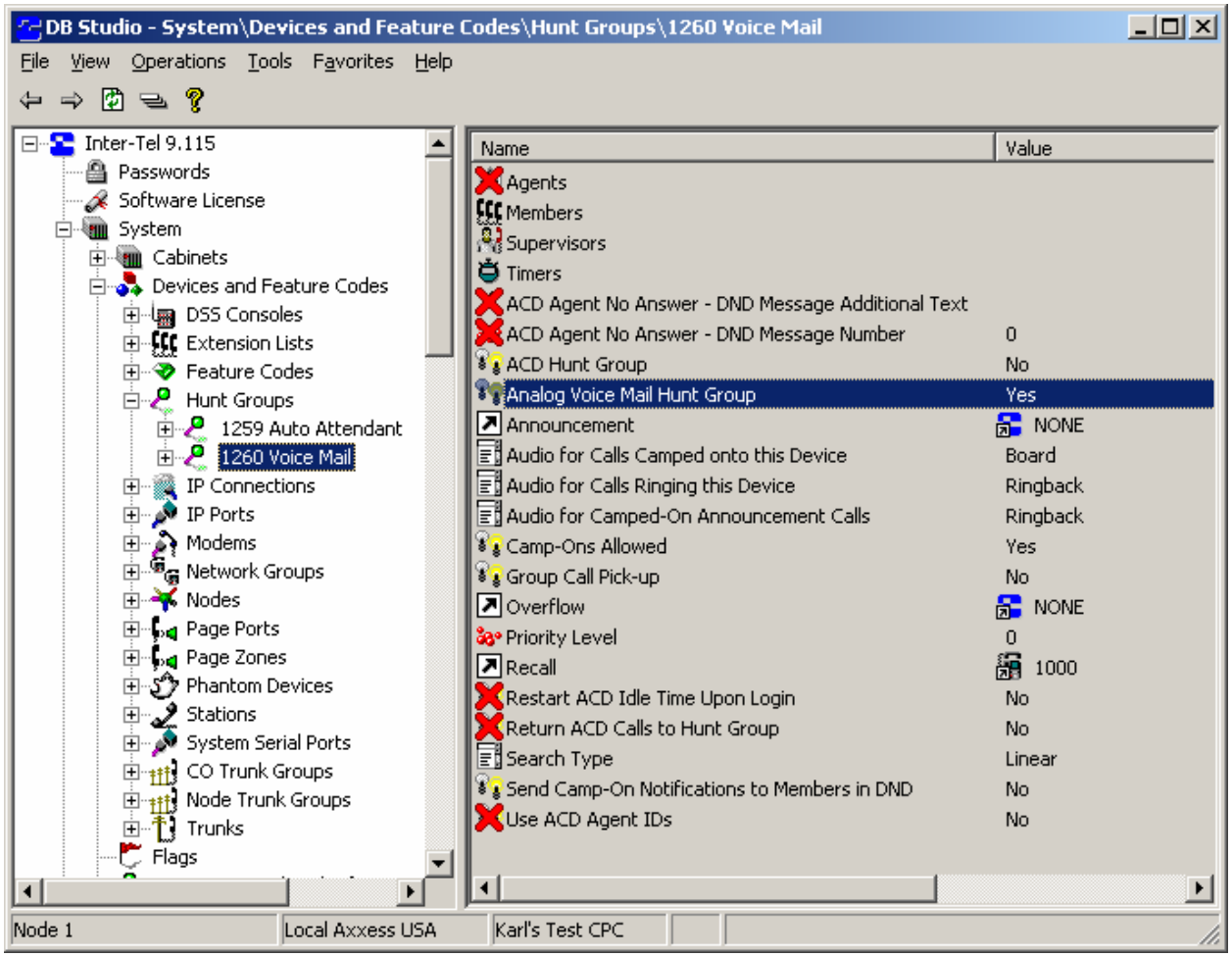


If there are no plans to implement Auto Attendant functionality, then all the OPX circuits can be placed in the Hunt Group for Voice Mail and vice versa.

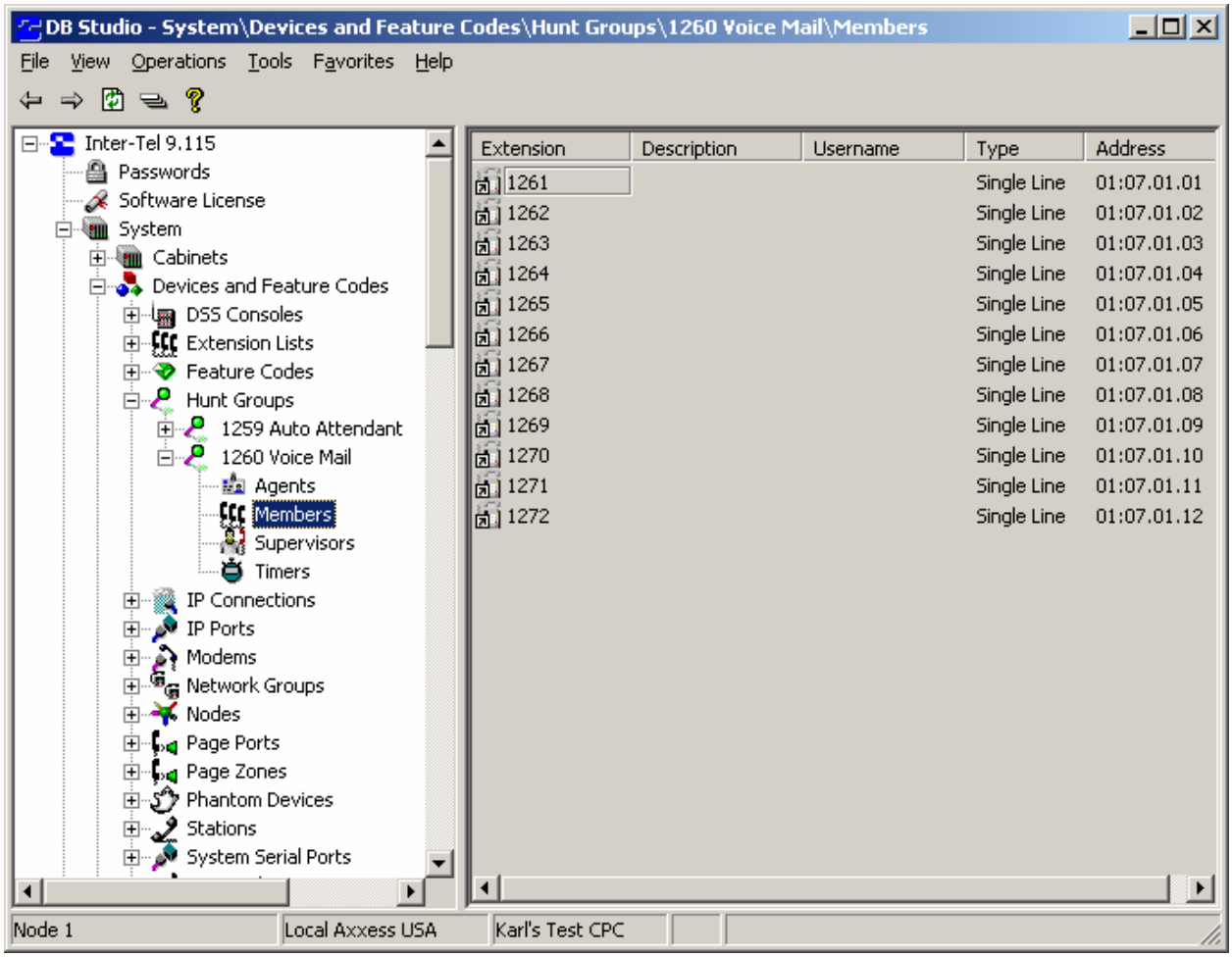
Note: When a call is placed to the Auto Attendant and is transferred back to the system, two circuits are used for the duration of the call. In addition, if the call is then forwarded to voice mail, a third circuit is used. The Audiocodes gateway bridges the audio between the trunks. **There is no route optimization**.

Step 5: Voice Mail Hunt Group Configuration

The hunt group used for voice mail should have the Analog Voice Mail Hunt Group flag set to "Yes."



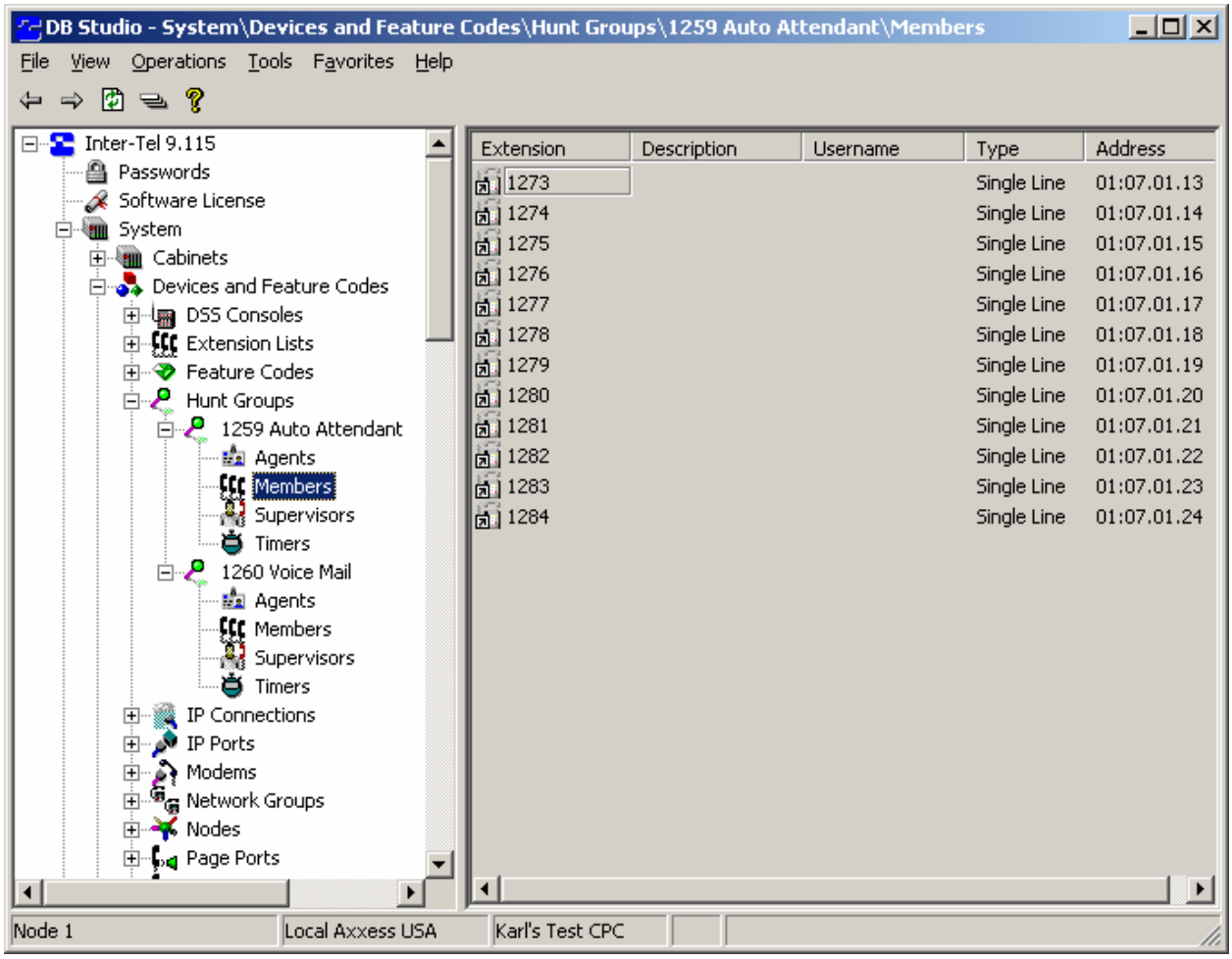
The Hunt Group Members list should include the first 'x' OPX circuits, where 'x' is the desired number of channels dedicated to voice mail access.



It's recommended that Camp-Ons Allowed remain enabled and that no Overflow destination be configured.

Step 6: Auto Attendant Hunt Group Configuration

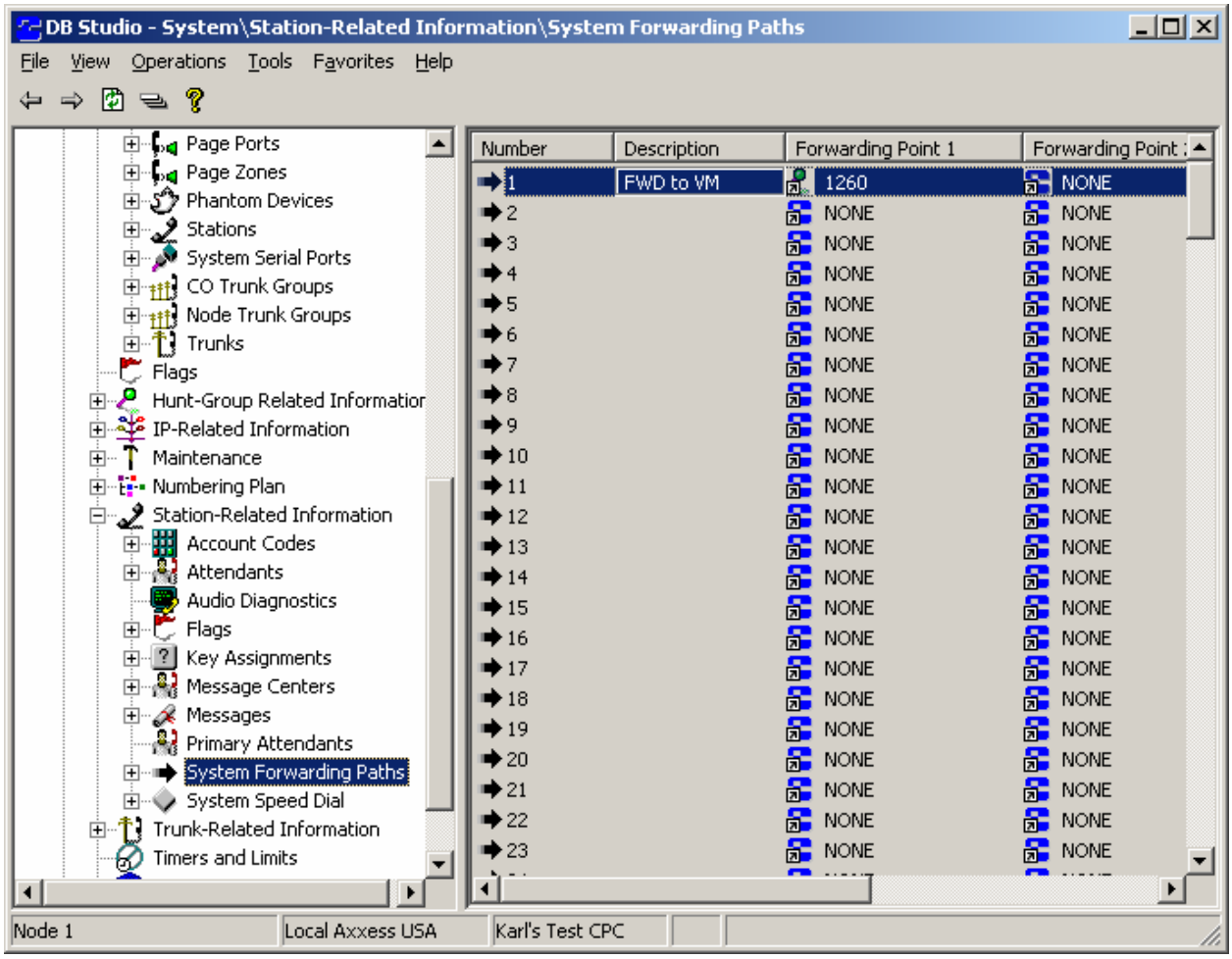
The hunt group used for Auto Attendant should leave the Analog Voice Mail Hunt Group flag set to "no."



The Hunt Group Members list should include the last 'x' OPX circuits, where 'x' is the remaining number of channels on the T1 dedicated to Exchange UM.

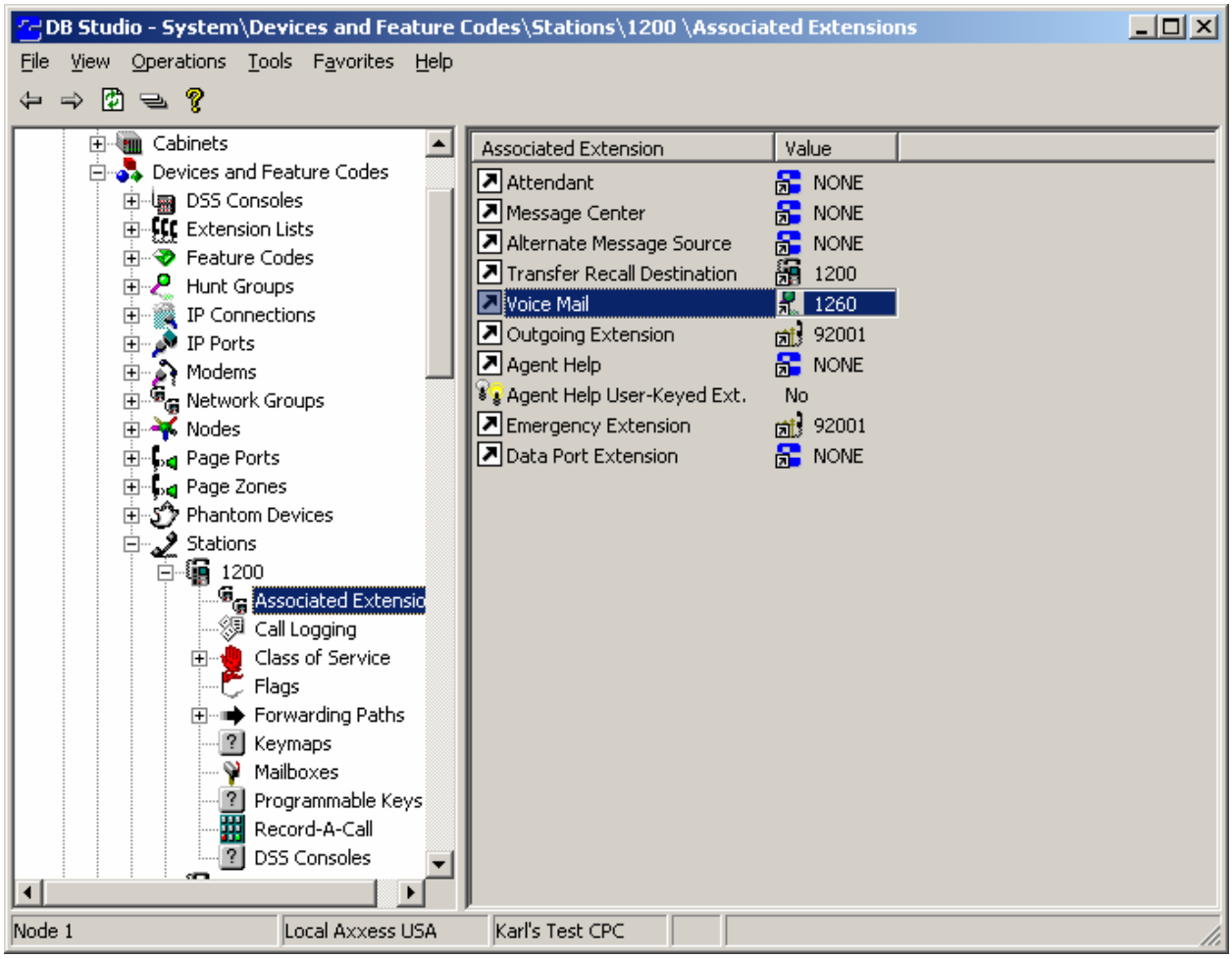
Step 7: System Forward Paths Configuration

Select an available System Forward Path and configure the first Forwarding Point to be the Voice Mail Hunt Group extension. Configure the Description to a descriptive name such as "FWD to VM".



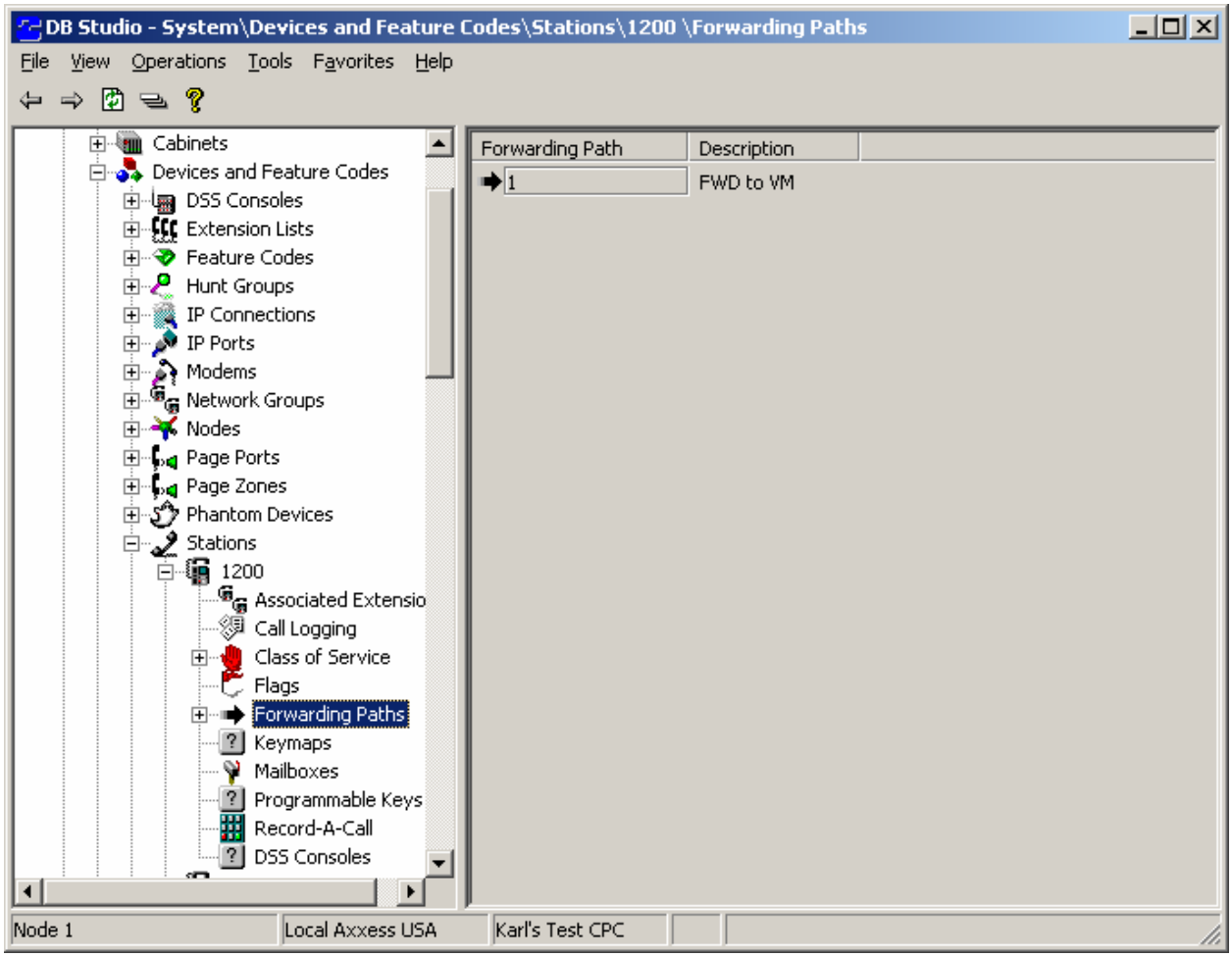
Step 8: Endpoint Configuration

For each endpoint, set the Voice Mail Associated Extension to the Voice Mail Hunt Group extension. It's recommended to use the station copy function for changing this on all endpoints.



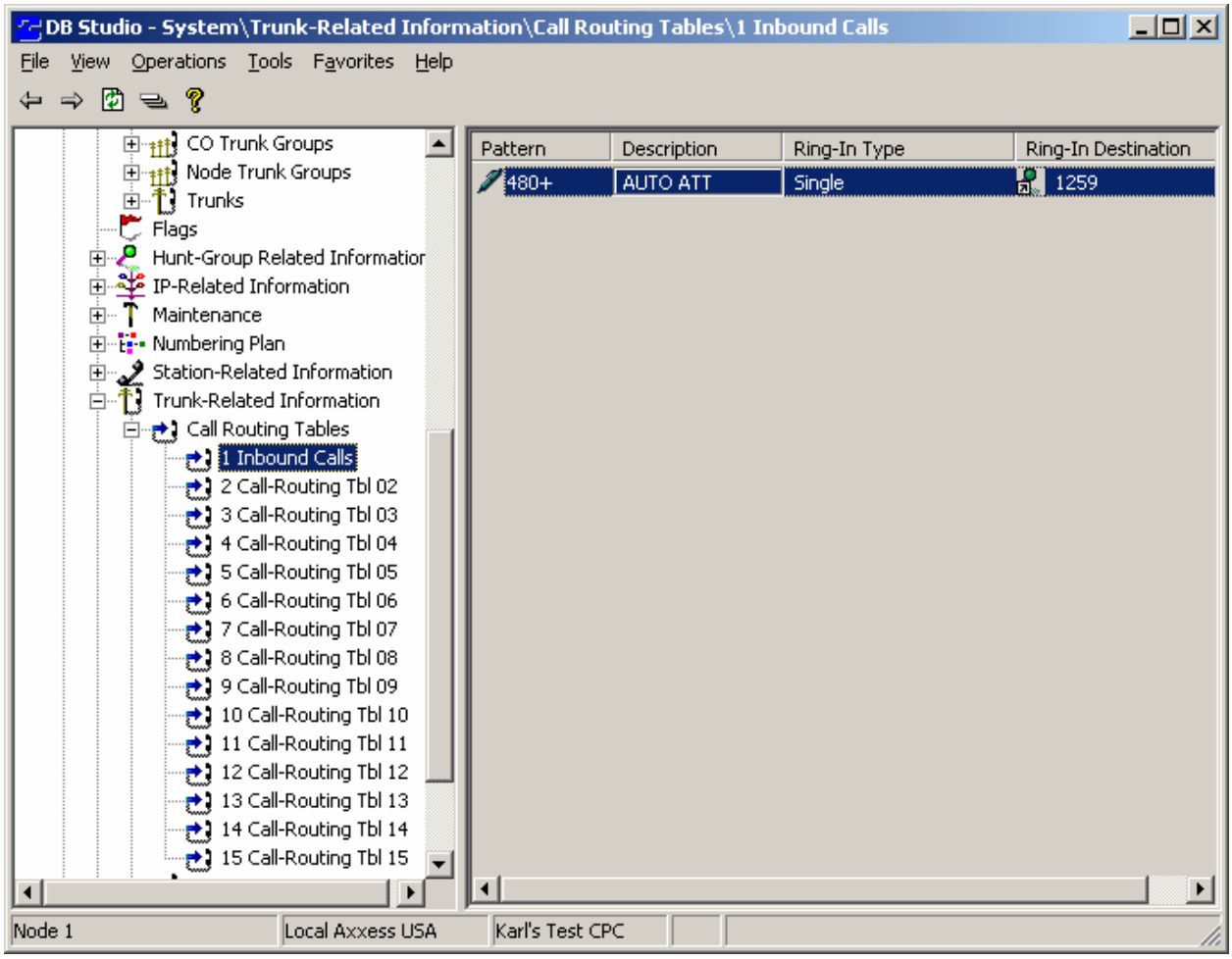
This enables the "Send Calls to Voice Mail" softkey on the 8660/8662 to redirect the caller to the Exchange UM Server.

For each endpoint, configure a Forwarding Path using the System Forward Path configuration described above. Set the conditions as appropriate for the site.



Step 9: Auto Attendant Configuration

To access the Exchange Auto Attendant, configure the Auto Attendant Hunt Group as the ring-in destination according to the call flows of the site.



5.1. TLS Setup

- N/A.

5.2. Fail-Over Configuration

- N/A.

5.3. Tested Phones

- Inter-Tel 8600 Series IP Endpoints
- Inter-Tel 8500 Series Digital Endpoints
- All digital endpoints supported by the Inter-Tel 5000 Series platforms

5.4. Other Comments

- N/A.

6. Exchange 2007 UM Validation Test Matrix

The following table contains a set of tests for assessing the functionality of the UM core feature set. The results are recorded as either:

- Pass (**P**)
- Conditional Pass (**CP**)
- Fail (**F**)
- Not Tested (**NT**)
- Not Applicable (**NA**)

Refer to:

- Appendix for a more detailed description of how to perform each call scenario.
- Section 6.1 for detailed descriptions of call scenario failures, if any.

No.	**Call Scenarios (see appendix for more detailed instructions)**	(**P**/**CP**/**F**/**NT**)	**Reason for Failure (see 6.1 for more detailed descriptions)**
1	Dial the pilot number from a phone extension that is NOT enabled for Unified Messaging and logon to a user's mailbox. Confirm hearing the prompt: "Welcome, you are connected to Microsoft Exchange. To access your mailbox, enter your extension…"	**P**	
2	Navigate mailbox using the Voice User Interface (VUI).	**P**	
3	Navigate mailbox using the Telephony User Interface (TUI).	**P**	
4	Dial user extension and leave a voicemail.		
4a	Dial user extension and leave a voicemail from an internal extension. Confirm the Active Directory name of the calling party is displayed in the sender field of the voicemail message.	**CP**	Inter-Tel 5000 PBX only sends the digits of the called user.
4b	Dial user extension and leave a voicemail from an external phone. Confirm the correct phone number of the calling party is displayed in the sender field of the voicemail message.	**CP**	Inter-Tel 5000 PBX only sends the digits of the called user.
5	Dial Auto Attendant (AA).	**P**	

	Dial the extension for the AA and confirm the AA answers the call.		
6	Call Transfer by Directory Search.		
6a	Call Transfer by Directory Search and have the called party answer. Confirm the correct called party answers the phone.	P	
6b	Call Transfer by Directory Search when the called party's phone is busy. Confirm the call is routed to the called party's voicemail.	P	
6c	Call Transfer by Directory Search when the called party does not answer. Confirm the call is routed to the called party's voicemail.	P	
6d	Setup an invalid extension number for a particular user. Call Transfer by Directory Search to this user. Confirm the number is reported as invalid.	P	
7	Outlook Web Access (OWA) Play-On-Phone Feature.		
7a	Listen to voicemail using OWA's Play-On-Phone feature to a user's extension.	P	
7b	Listen to voicemail using OWA's Play-On-Phone feature to an external number.	P	
8	Configure a button on the phone of a UM-enabled user to forward the user to the pilot number. Press the voicemail button. Confirm you are sent to the prompt: "Welcome, you are connected to Microsoft Exchange. <User>. Please enter your pin and press the pound key."	CP	Unable to perform direct subscriber access. The user is prompted to first enter the mailbox number, and then the pin number.
9	Send a test FAX message to user extension. Confirm the FAX is received in the user's inbox.	P	
10	Setup TLS between gateway/IP-PBX and		

	Exchange UM. Replace this italicized text with your TLS configuration: self-signed certificates or Windows Certificate Authority (CA).		
10a	Dial the pilot number and logon to a user's mailbox. Confirm UM answers the call and confirm UM responds to DTMF input.	**NT**	The gateway supports TLS. However, TLS implementation is currently being tested.
10b	Dial a user extension and leave a voicemail. Confirm the user receives the voicemail.	**NT**	The gateway supports TLS. However, TLS implementation is currently being tested.
10c	Send a test FAX message to user extension. Confirm the FAX is received in the user's inbox.	**NT**	The gateway supports TLS. However, TLS implementation is currently being tested.
11	Setup G.723.1 on the gateway. (If already using G.723.1, setup G.711 A Law or G.711 Mu Law for this step). Dial the pilot number and confirm the UM system answers the call.	**P**	
12	Setup Message Waiting Indicator (MWI). Geomant offers a third party solution: MWI 2007. Installation files and product documentation can be found on Geomant's MWI 2007 website.	**NT**	
13	Execute Test-UMConnectivity.	**NT**	
14	Setup and test fail-over configuration on the IP-PBX to work with two UM servers.	**NA**	

6.1. Detailed Description of Limitations

Failure Point	Inter-Tel 5000 PBX sends only the digits of the called user.
Phone type (if phone-specific)	All types
Call scenarios(s) associated with failure point	4a, 4b
List of UM features affected by failure point	The calling user number is not displayed in the sender field of the voice mail message.
Additional Comments When a call is forwarded to the voice mail, the Inter-Tel 5000 PBX sends only the digits of the called user without information about the calling number. As a result, when the calling user leaves a voice mail message its calling number is not displayed in the sender field of the voice mail message.	

Failure Point	The Inter-Tel 5000 doesn't support the sending of the calling number.
Phone type (if phone-specific)	All Types
Call scenarios(s) associated with failure point	8.
List of UM features affected by failure point	Direct call to the Exchange 2007 UM Server.
Additional Comments The Inter-Tel 5000 don't support the sending of the calling user number for direct calls (i.e., user calls UM to retrieve voice message). Therefore, when the user dials directly to the Exchange 2007 UM Server, the user hears the general welcome prompt: "Welcome, you are connected to Microsoft Exchange, to access your mailbox, enter your extension.", at which the user is required to enter the user's extension number in addition to the pin number.	

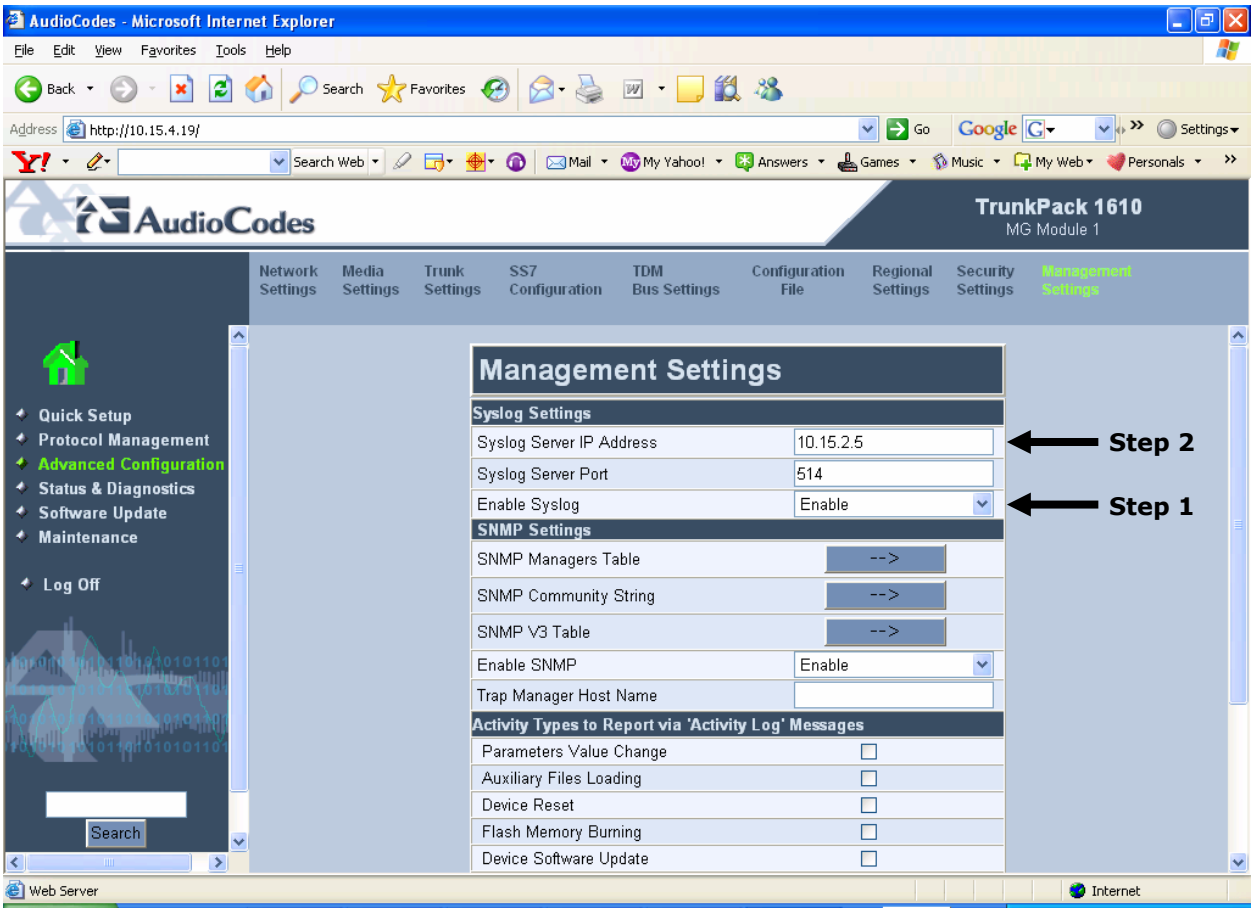
7. Troubleshooting

The tools used for debugging include network sniffer applications (such as Ethereal) and AudioCodes' Syslog protocol.

The Syslog client, embedded in the AudioCodes gateways (MP-11x, Mediant 1000, and Mediant 2000), sends error reports/events generated by the gateway application to a Syslog server, using IP/UDP protocol.

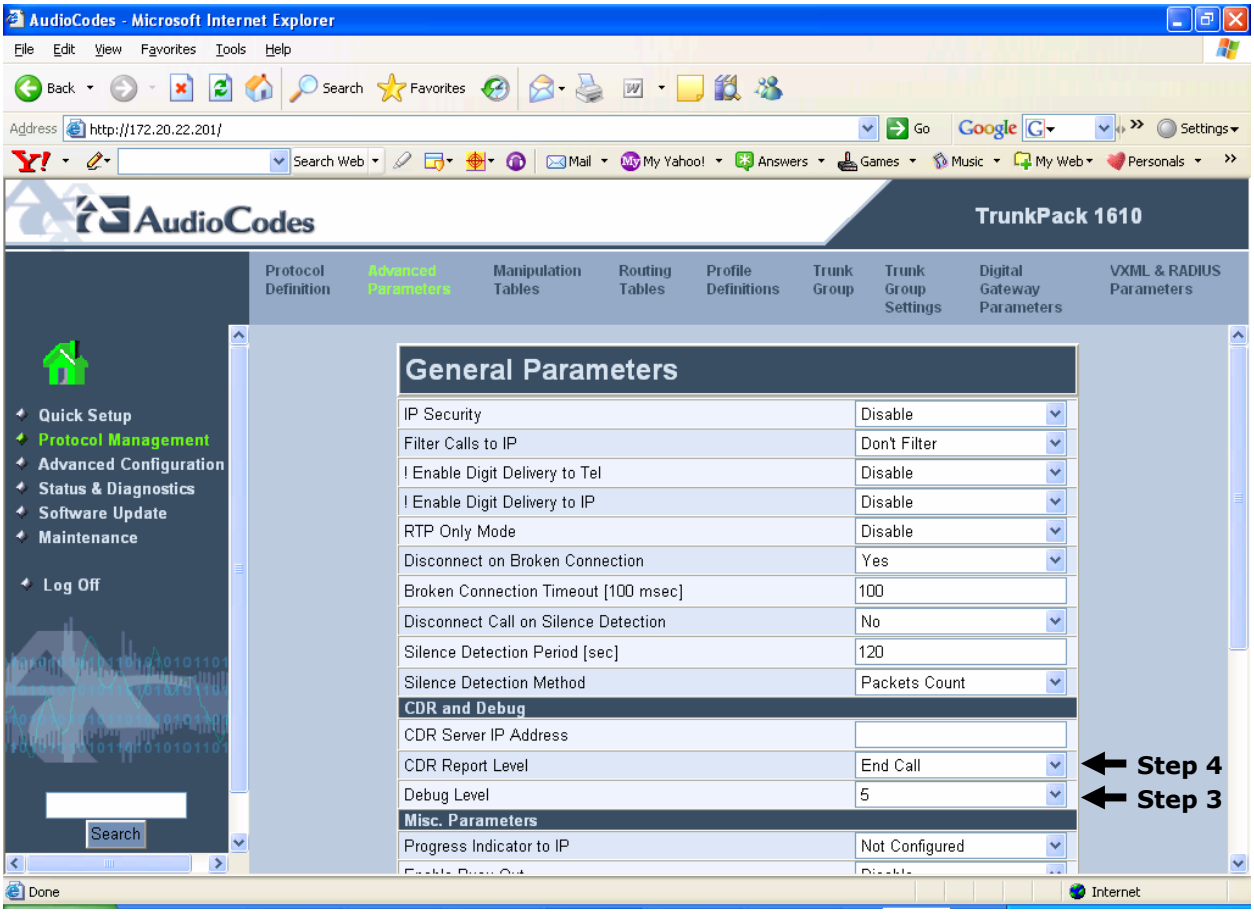
To activate the Syslog client on the AudioCodes gateways:

1. Set the parameter **Enable Syslog** to 'Enable'.
2. Use the parameter **Syslog Server IP Address** to define the IP address of the Syslog server you use.



Note: The Syslog Server IP address must be one that corresponds to your network environment in which the Syslog server is installed (for example, 10.15.2.5).

3. To determine the Syslog logging level, use the parameter **Debug Level** and set this parameter to '5'.
4. Change the **CDR Report Level** to 'End Call' to enable additional call information.



AudioCodes has also developed advanced diagnostic tools that may be used for high-level troubleshooting. These tools include the following:

- PSTN Trace: PSTN Trace is a procedure used to monitor and trace the PSTN elements (E1/T1) in AudioCodes digital gateways (Mediant 1000 & Mediant 2000). These utilities are designed to convert PSTN trace binary files to textual form.
- DSP Recording: DSP recording is a procedure used to monitor the DSP operation (e.g., rtp packets and events).

Appendix

1. Dial Pilot Number and Mailbox Login

- Dial the pilot number of the UM server from an extension that is NOT enabled for UM.
- Confirm hearing the greeting prompt: "Welcome, you are connected to Microsoft Exchange. To access your mailbox, enter your extension..."
- Enter the extension, followed by the mailbox PIN of an UM-enabled user.
- Confirm successful logon to the user's mailbox.

2. Navigate Mailbox using Voice User Interface (VUI)

- Logon to a user's UM mailbox.
- If the user preference has been set to DTMF tones, activate the Voice User Interface (VUI) under personal options.
- Navigate through the mailbox and try out various voice commands to confirm that the VUI is working properly.
- This test confirms that the RTP is flowing in both directions and speech recognition is working properly.

3. Navigate Mailbox using Telephony User Interface (TUI)

- Logon to a user's UM mailbox.
- If the user preference has been set to voice, press "#0" to activate the Telephony User Interface (TUI).
- Navigate through the mailbox and try out the various key commands to confirm that the TUI is working properly.
- This test confirms that both the voice RTP and DTMF RTP (RFC 2833) are flowing in both directions.

4. Dial User Extension and Leave Voicemail

- Note: If you are having difficulty reaching the user's UM voicemail, verify that the coverage path for the UM-enabled user's phone is set to the pilot number of the UM server.

a. From an Internal Extension

- From an internal extension, dial the extension for a UM-enabled user and leave a voicemail message.
- Confirm the voicemail message arrives in the called user's inbox.
- Confirm this message displays a valid Active Directory name as the sender of this voicemail.

b. From an External Phone

- From an external phone, dial the extension for a UM-enabled user and leave a voicemail message.
- Confirm the voicemail message arrives in the called user's inbox.
- Confirm this message displays the phone number as the sender of this voicemail.

5. Dial Auto Attendant(AA)

- Create an Auto Attendant using the Exchange Management Console:
  - Under the Exchange Management Console, expand "Organizational Configuration" and then click on "Unified Messaging".
  - Go to the Auto Attendant tab under the results pane.
  - Click on the "New Auto Attendant…" under the action pane to invoke the AA wizard.
  - Associate the AA with the appropriate dial plan and assign an extension for the AA.
  - Create PBX dialing rules to always forward calls for the AA extension to the UM server.
  - Confirm the AA extension is displayed in the diversion information of the SIP Invite.
- Dial the extension of Auto Attendant.
- Confirm the AA answers the call.

6. Call Transfer by Directory Search

- Method one: Pilot Number Access
  - Dial the pilot number for the UM server from a phone that is NOT enabled for UM.
  - To search for a user by name:
  - Press # to be transferred to name Directory Search.
    - Call Transfer by Directory Search by entering the name of a user in the same Dial Plan using the telephone keypad, last name first.
  - To search for a user by email alias:
    - Press "#" to be transferred to name Directory Search
    - Press "# #" to be transferred to email alias Directory Search
    - Call Transfer by Directory Search by entering the email alias of a user in the same Dial Plan using the telephone keypad, last name first.
- Method two: Auto Attendant
  - Follow the instructions in appendix section 5 to setup the AA.
  - Call Transfer by Directory Search by speaking the name of a user in the same Dial Plan. If the AA is not speech enabled, type in the name using the telephone keypad.
- Note: Even though some keys are associated with three or four numbers, for each letter, each key only needs to be pressed once regardless of the letter you want. Ignore spaces and symbols when spelling the name or email alias.

a. Called Party Answers

- Call Transfer by Directory Search to a user in the same dial plan and have the called party answer.
- Confirm the call is transferred successfully.

b. Called Party is Busy

- Call Transfer by Directory Search to a user in the same dial plan when the called party is busy.
- Confirm the calling user is routed to the correct voicemail.

c. Called Party does not Answer

- Call Transfer by Directory Search to a user in the same dial plan and have the called party not answer the call.
- Confirm the calling user is routed to the correct voicemail.

d. The Extension is Invalid

- Assign an invalid extension to a user in the same dial plan. An invalid extension has the same number of digits as the user's dial plan and has not been mapped on the PBX to any user or device.
    - UM Enable a user by invoking the "Enable-UMMailbox" wizard.
    - Assign an unused extension to the user.
    - Do not map the extension on the PBX to any user or device.
    - Call Transfer by Directory Search to this user.
    - Confirm the call fails and the caller is prompted with appropriate messages.

7. Play-On-Phone

- To access play-on-phone:
    - Logon to Outlook Web Access (OWA) by going to URL https://<server name>/owa.
    - After receiving a voicemail in the OWA inbox, open this voicemail message.
    - At the top of this message, look for the Play-On-Phone field (Play on Phone...).
    - Click this field to access the Play-On-Phone feature.

a. To an Internal Extension

- Dial the extension for a UM-enabled user and leave a voicemail message.
- Logon to this called user's mailbox in OWA.
- Once it is received in the user's inbox, use OWA's Play-On-Phone to dial an internal extension.
- Confirm the voicemail is delivered to the correct internal extension.

b. To an External Phone number

- Dial the extension for a UM-enabled user and leave a voicemail message.
- Logon to the UM-enabled user's mailbox in OWA.

- Confirm the voicemail is received in the user's mailbox.
- Use OWA's Play-On-Phone to dial an external phone number.
- Confirm the voicemail is delivered to the correct external phone number.
- Troubleshooting:
    - Make sure the appropriate UMMailboxPolicy dialing rule is configured to make this call. As an example, open an Exchange Management Shell and type in the following commands:
    - $dp = get-umdialplan -id <dial plan ID>
    - $dp.ConfiguredInCountryOrRegionGroups.Clear()
    - $dp.ConfiguredInCountryOrRegionGroups.Add("anywhere,*,*,")
    - $dp.AllowedInCountryOrRegionGroups.Clear()
    - $dp.AllowedInCountryOrRegionGroups.Add("anywhere")
    - $dp|set-umdialplan
    - $mp = get-ummailboxpolicy -id <mailbox policy ID>
    - $mp.AllowedInCountryGroups.Clear()
    - $mp.AllowedInCountryGroups.Add("anywhere")
    - $mp|set-ummailboxpolicy
    - The user must be enabled for external dialing on the PBX.
    - Depending on how the PBX is configured, you may need to prepend the trunk access code (e.g. 9) to the external phone number.

8. Voicemail Button

- Configure a button on the phone of a UM-enabled user to route the user to the pilot number of the UM server.
- Press this voicemail button on the phone of an UM-enabled user.
- Confirm you are sent to the prompt: "Welcome, you are connected to Microsoft Exchange. <User Name>. Please enter your pin and press the pound key."
- Note: If you are not hearing this prompt, verify that the button configured on the phone passes the user's extension as the redirect number. This means that the user extension should appear in the diversion information of the SIP invite.

9. FAX

- Use the Management Console or the Management Shell to FAX-enable a user.
- Management Console:
    - Double click on a user's mailbox and go to Mailbox Features tab.
    - Click Unified Messaging and then click the properties button.
    - Check the box "Allow faxes to be received".
- Management Shell - execute the following command:
    - Set-UMMailbox –identity UMUser –FaxEnabled:$true

- To test fax functionality:
    - Dial the extension for this fax-enabled UM user from a fax machine.
    - Confirm the fax message is received in the user's inbox.
    - Note: You may notice that the UM server answers the call as though it is a voice call (i.e. you will hear: "Please leave a message for…"). When the UM server detects the fax CNG tones, it switches into fax receiving mode, and the voice prompts terminate.
    - Note: UM only support T.38 for sending fax.

10. TRANSPORT SECURITY LAYER (TLS)

- Setup TLS on the gateway/IP-PBX and Exchange 2007 UM.
- Import/Export all the appropriate certificates.

a. Dial Pilot Number and Mailbox Login

- Execute the steps in scenario 1 (above) with TLS turned on.

b. Dial User Extension and Leave a Voicemail

- Execute the steps in scenario 4 (above) with TLS turned on.

c. FAX

- Execute the steps in scenario 9 (above) with TLS turned on.

11. G.723.1

- Configure the gateway to use the G.723.1 codec for sending audio to the UM server.
- If already using G.723.1 for the previous set of tests, use this step to test G.711 A Law or G.711 Mu Law instead.
- Call the pilot number and verify the UM server answers the call.
- Note: If the gateway is configured to use multiple codecs, the UM server, by default, will use the G.723.1 codec if it is available.

12. Message Waiting Indicator (MWI)

- Although Exchange 2007 UM does not natively support MWI, Geomant has created a 3rd party solution - MWI2007. This product also supports SMS message notification.
- Installation files and product documentation can be found on Geomant's MWI 2007 website.

13. Test-UMConnectivity

- Run the Test-UMConnectivity diagnostic cmdlet by executing the following command in Exchange Management Shell:
- Test-UMConnectivity –UMIPGateway:<Gateway> -Phone:<Phone> |fl
- <Gateway> is the name (or IP address) of the gateway which is connected to UM, and through which you want to check the connectivity to the UM server. Make sure the gateway is configured to route calls to UM.
- <Phone> is a valid UM extension. First, try using the UM pilot number for the hunt-group linked to the gateway. Next, try using a CFNA number configured for the gateway. Please ensure that a user or an AA is present on the UM server with that number.
- The output shows the latency and reports if it was successful or there were any errors.

14. Test Fail-Over Configuration on IP-PBX with Two UM Servers

- This is only required for direct SIP integration with IP-PBX. If the IP-PBX supports fail-over configuration (e.g., round-robin calls between two or more UM servers):
    - Provide the configuration steps in Section 5.
    - Configure the IP-PBX to work with two UM servers.
    - Simulate a failure in one UM server.
    - Confirm the IP-PBX transfers new calls to the other UM server successfully.